1996

An Integrated Network Architecture for a High Speed Distributed Multimedia System.

Nitin Subrao Naik
Louisiana State University and Agricultural & Mechanical College

AN INTEGRATED NETWORK ARCHITECTURE FOR A HIGH SPEED DISTRIBUTED MULTIMEDIA SYSTEM

A Dissertation

Submitted to the Graduate Faculty of the
Louisiana State University and
Agricultural and Mechanical College
in partial fulfillment of the
requirements for the degree of
Doctor of Philosophy

in

The Department of Computer Science

by
Nitin S. Naik
B.E., University of Mysore, 1984
M.S.E.E., Louisiana State University, 1988
M.S.S.S., Louisiana State University, 1990
August 1996

UMI Number: 9706351

Acknowledgement

I would like to express my earnest gratitude to my advisor, Professor Sitharama Iyengar, for motivating me to pursue a Ph.D. and for the essential guidance provided throughout this research. I would also like to thank the members of my doctoral committee, Professors Bert Boyce, Bush Jones, Donald Kraft and Gerald Knapp for carefully reading my dissertation and providing suggestions for improvement.

I would like to thank the staff at the NASA Classroom of the Future Program, especially Drs. Daniel Miller and Craig Blurton for their constant support and encouragement during this research. I am also grateful to Bruce Rosen, Ron Magers, Tom Stoffel and Greg Sydor for all their support in implementing this research project. I would also like to thank the team at Bay Networks for enabling me put this network together and especially Brian Smith, Bryan Petty, Kevin Parks, Randy Uram and David Cohen in making available all the tools necessary to collect data and verify the design.

I am also grateful to Dr. Eugene and Jeanne Tims, and Dr. Chad and Bert Ellett for providing constant support, motivation and for being on my side for every step of the way. They have made my stay at L.S.U and in the U.S. a very happy and memorable one.

Special thanks and appreciation for my wife Priyali who gave a new meaning to sacrifice, love and being by my side all through the trials and tribulations of our lives. Priyali's parents and sister have also provided support and encouragement throughout my doctoral studies, which is greatly appreciated. Finally, although my parents and sister are on the other side of the Atlantic, they have been a constant source of encouragement, support and vision not only during my education but throughout my life, for which I consider myself fortunate and extend my heartfelt gratitude.

Table of Contents

List of Tables

List of Figures

List of Acronyms

AAL	ATM Adaptation Layer
A/D	Analog-to-Digital
ADPCM	Adaptive Differential Pulse Code Modulation
ATDM	Asynchronous Time Division Multiplexing
ATM	Asynchronous Transfer Mode
B-ISDN	Broadband Integrated Services Digital Networks
CBR	Constant Bit Rate
CD	Compact Disk
CET	Center for Educational Technologies
CCITT	Committée Consultatif International Télégraphique et Téléphonique
CS	Convergence Sublayer
CSMA-CD	Carrier Sense Multiple Access - Collision Detection
CODEC	COder-DECoder
COTF	Classroom of the Future Program
C/V LAN	Computer/Video Local Area Network
DAN	Desk Area Network
DCT	Discrete Cosine Transformation
DPCM	Differential Pulse Code Modulation
DVI	Digital Video Interactive
FDDI	Fiber Distributed Data Interface

FEC	Forward Error Correction
GB	Gigabytes
Gbps	Gigabits per second
HDTV	High Definition Television
I/O	Input-Output
ISDN	Integrated Services Digital Network
ISO	International Standards Organization
JPEG	Joint Photographic Experts Group
KB	Kilobytes
Kbps	Kilobits per second
KHz	KiloHertz
LAN	Local Area Network
LDU	Logical Data Unit
MAC	Medium Access Control
MAN	Metropolitan Area Network
MB	Megabytes
Mbps	Megabits per second
MOS	Multimedia Operating System
MPEG	Motion Picture Experts Group
MTP	Multimedia Transport Protocol
NASA	National Aeronautics and Space Agency
NII	National Information Infrastructure

NLM	Network Loadable Module
NSF	National Science Foundation
NTSC	National Television Standards Committee
OSI	Open Systems Interconnection
PCM	Pulse Code Modulation
PDU	Protocol Data Unit
QoS	Quality of Service
RAM	Random Access Memory
RFC	Request For Comments
ROM	Read Only Memory
SAR	Segmentation and Reassembly
SDU	Session Data Unit
SONET	Synchronous Optical Network
ST-II	Stream Protocol version II
STM	Synchronous Transfer Mode
TV	Television
VBR	Variable Bit Rate
VC	Virtual Channel
VCR	Video Cassette Recorder
VP	Virtual Path
WAN	Wide Area Network

Abstract

Computer communication demands for higher bandwidth and smaller delays are increasing rapidly as the march into the twenty-first century gains momentum. These demands are generated by visualization applications which model complex real time phenomena in visual form, electronic document imaging and manipulation, concurrent engineering, on-line databases and multimedia applications which integrate audio, video and data. The convergence of the computer and video worlds is leading to the emergence of a distributed multimedia environment.

This research investigates an integrated approach in the design of a high speed computer-video local area network for a distributed multimedia environment. The initial step in providing multimedia services over computer networks is to ensure bandwidth availability for these services. The bandwidth needs based on traffic generated in a distributed multimedia environment is computationally characterized by a model. This model is applied to the real-time problem of designing a backbone for a distributed multimedia environment at the NASA Classroom of the Future Program. The network incorporates legacy LANs and the latest high speed switching technologies. Performance studies have been conducted with different network topologies for various multimedia application scenarios to establish benchmarks for the operation of the network. In these performance studies it has been observed that network topologies play an

important role in ensuring that sufficient bandwidth is available for multimedia traffic.

After the implementation of the network and the performance studies, it was found that for true quality of service guarantees, some modifications will have to be made in the multimedia operating systems used in client workstations. These modifications would gather knowledge of the channel between source and destination and reserve resources for multimedia communication based on specified requirements. A scheme for reserving resources in a network consisting legacy LAN and ATM is presented to guarantee quality of service for multimedia applications.

Chapter 1

Introduction

1.1 Perspective

A number of trends today are leading to an accelerating need for ever higher network bandwidth. The power of desktop platforms is increasing at a very rapid pace, as personal computers become faster and workstation prices continue to decline. In order to exploit the growing power, application developers and leading edge users are looking today at entirely new classes of client-server applications. These include visualization applications which model complex real time phenomena in visual form, electronic document imaging and manipulation, concurrent engineering, on-line databases, and multimedia applications which integrate voice, video and data. A common trend underlying such applications is the growing use of interactive visual imagery to facilitate communication and the processing and presentation of complex data.

Hence, as the march into the twenty-first century gains momentum, computer communication demands for higher bandwidth and smaller delays are increasing rapidly. These demands are generated, as indicated above, with the migration of applications from specialized networks to the common network such as the Internet. This has also lead to a serious debate on a National Information Infrastructure (NII) for the U.S. This discussion has also involved international entities where Internet connectivity is possible.

A review of the NSF research priorities [36] indicates that the communication network architectures suitable for the NII will be required to support a wide range of applications, in a cost-effective manner, with unprecedented reliability and on a very large scale. The applications will vary from data rates of a few bits per second to gigabits per second or the applications may have data rate demands that may vary widely during a short time period. The network architectures must provide a range of quality-of-service options to accommodate different performance and cost requirements. The network must also support one-to-many and many-to-many communication channels to support information distribution and collaborative applications. In addition, the NII will require network architectures that can accommodate evolution across multiple generations of technology and that facilitate heterogeneity in applications, end systems, transmission technologies and switching mechanisms. To allow future networks to be managed effectively they must provide mechanisms for traffic measurement, error monitoring, usage accounting and cost recovery. The research presented in this dissertation investigates one possible network configuration for client-server based multimedia applications.

1.2 Motivation for the Study

Besides the need for high speed networks, the demand for convergence of the computer and video worlds is seeing rapid growth in the last few years. The applications generating these demands are from the medical community, cable television providers, education institutions, training personnel, financial decision makers, large retailers, etc. These audiences recognize that the effectiveness of communication increases with the integration of text, graphics, images, sound, animation and video. This integration has been termed multimedia.

In the past, this integration has been accomplished by the use of separate computer and video networks. The computer network would carry the text, graphics and maybe images to the user's computer monitor while the video network or system would provide the sound, animation, images and video on a separate video monitor or TV. The only connection between the two being the control cable from the computer to the video system as shown in Figure 1.1.

At the present time, due to the research and development of enabling technologies in the field of digital video, faster computer workstations and servers, multi-sync monitors, network technologies, analog-to-digital convertors, multimedia extensions in the operating systems, etc. a true integration of the computer-video network is possible as shown in Figure 1.2.

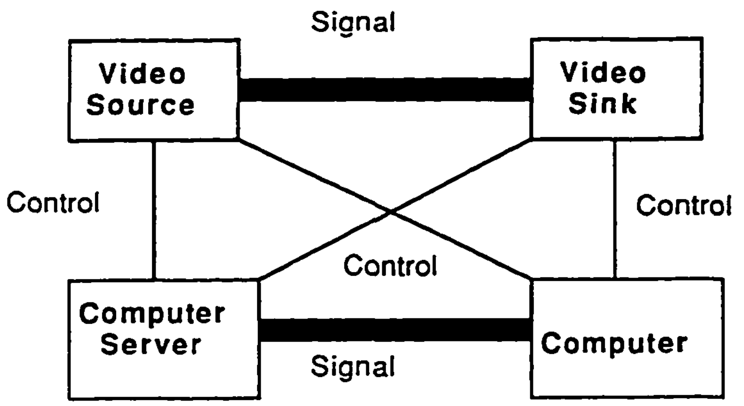


Figure 1.1 : Separate Video Network and Computer Networks

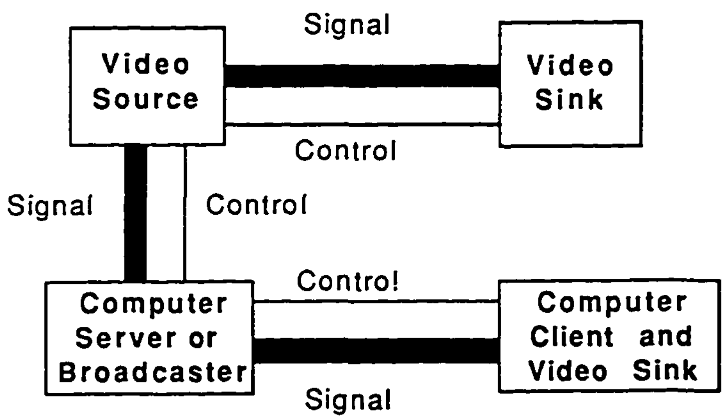


Figure 1.2 : Integrated Computer and Video Network

When this desire for an integrated network is coupled with the need for a mechanism to share multimedia applications in our present client-server world, it gives rise to the distributed multimedia system paradigm. Distributed multimedia systems require a network architecture in which elements such as devices, schedules and protocol stacks are coordinated as a system [46]. But, managing multiple devices with different data and file formats adds considerable complexity to the network and the operating system. Each device has different requirements and allowable trade-offs, which could conflict with each other for a given application. Further, the operation of the distributed system is delay sensitive and hence must employ services that provide timing guarantees. The system needs can be met if the time and resource requirements are communicated to the various components and updated during the execution of an application. These requirements for a quality of service (QoS) desired by the application, are represented by the following parameters [20]: (a) application QoS parameters which relate to the settings for multimedia devices; (b) system QoS parameters which relate to the operating system services; and (c) network QoS parameters which relate to the network resources. It must be noted that though the behavior of these components is interdependent in a distributed multimedia system, from an application perspective, network resource management plays a very important role.

Resource orchestration plays a central role in maintaining the stability of a distributed multimedia system [41]. The resources exist at the endpoints which are hosts or clients and at the network switches or intermediate network nodes. The resources which are found at the input or output endpoints are represented with the following application QoS parameters; specific media characteristics such as sample size (height, width and color specification in a video stream); transmission characteristics requirements for end-to-end delivery (delay bounds), communication topology (point-to-point or point-to-multipoint), and media relations (synchronization).

The network QoS parameters can be divided into three classes [47]. The first class covers basic parameters such as packet size and packet rate. In the second, are environment sensitive parameters such as interarrival delay, end-to-end delay and packet loss rates. The third class specifies overall communication requirements that are represented by parameters such as packet ordering, communication topology, cost and transmission priorities.

Operating system resources include processing times required for tasks, secondary storage and memory buffer requirements. The operating system resources, which are represented by system parameters, are used to match the application QoS requirements to the network QoS parameters.

1.3 Scope of this Dissertation

In order to understand the parameters to be considered for reliable network operation in a distributed multimedia system, it is necessary to construct a system which integrates the video and computer worlds using traditional off-the-shelf products. This system must operate not only under normal client-server situations with traditional wordprocessing, spreadsheet, database and mail applications, but also with client-server type multimedia applications and one-to-one or one-to-many video conferences. As part of this dissertation work, an integrated computer-video LAN prototype was established at the Center for Educational Technologies that houses the NASA Classroom of the Future program.

Based on this implementation, the contributions of this dissertation are:

1. A traffic model to determine the capacity of an integrated network architecture for a distributed multimedia system.
2. A model for determining the size of various components in the multimedia server architecture.
3. An integrated design of a computer-video local area network based on legacy LAN and new switching technologies.
4. Benchmarks for the operation of the network for client-server based multimedia applications and video conference broadcasts.
5. A framework for establishing channels between source and destination that can guarantee quality of service in the network consisting of legacy LAN and ATM technologies for multimedia communication.

1.4 Overview

The dissertation concentrates on the design, implementation and analysis of an integrated network architecture that combines the computer and video networks seamlessly in a distributed multimedia system.

Chapter 2 deals with the background of terms and concepts that one must be familiar with when reading about this topic. It also presents other projects found in the literature with the goal of establishing a stable distributed multimedia system.

Chapter 3 describes the requirements that need to be considered when designing an integrated multimedia system. A detailed discussion is presented on a mathematical model for determining bandwidth requirements for the network and capacity calculations for the multimedia server. Issues such as incorporation of legacy LANs, scalability, error recovery and transmission priorities are investigated in detail.

Chapter 4 describes the distributed multimedia environment implemented at the Center for Educational Technologies for the NASA Classroom of the Future Program and presents the results of performance studies conducted on the network using the various components of the environment.

A new scheme for the establishment of network channels for guaranteed quality of service is presented in chapter 5.

Chapter 6 contains conclusions of the research of the distributed multimedia system and discussions about some open research problems.

Chapter 2

Background

2.1 Preliminaries

The design of a network architecture requires some understanding of data communications and the Open System Interconnection (OSI) model. If the network architecture is used in a distributed multimedia environment, then we must gain a full understanding of file formats, data structures, computing and communication requirements for multimedia. In this chapter we will study the above topics in detail.

2.1.1 Data Communications

Data communications can be considered as the transfer of information from one computing element to another. The type of traffic being carried defines the requirements of the communications medium and the access methodology [39]. Data files embrace many kinds of information. They may be computer programs, graphic files, stored video files for later playback, application program data files, and so on. The need for higher bandwidth and real-time support depends on the requirements of the application programs involved and the latency that the user can tolerate. Both are typically needed for the transmission of files between server systems and end stations.

Transaction processing is another application for wideband shared local area networks (LAN) or dedicated point-to-point connections of moderate bandwidth (Figure 2.1). This job usually involves transferring single messages containing only small sets of application-specific information. It has been shown that interactive computer applications such as data entry are most effective when screen updates (usually just text) should occur within half a second. Then the system appears to the user to be functioning in real time. Summarizing, transaction processing involves the fast transfer of information but not much of it at any one time.

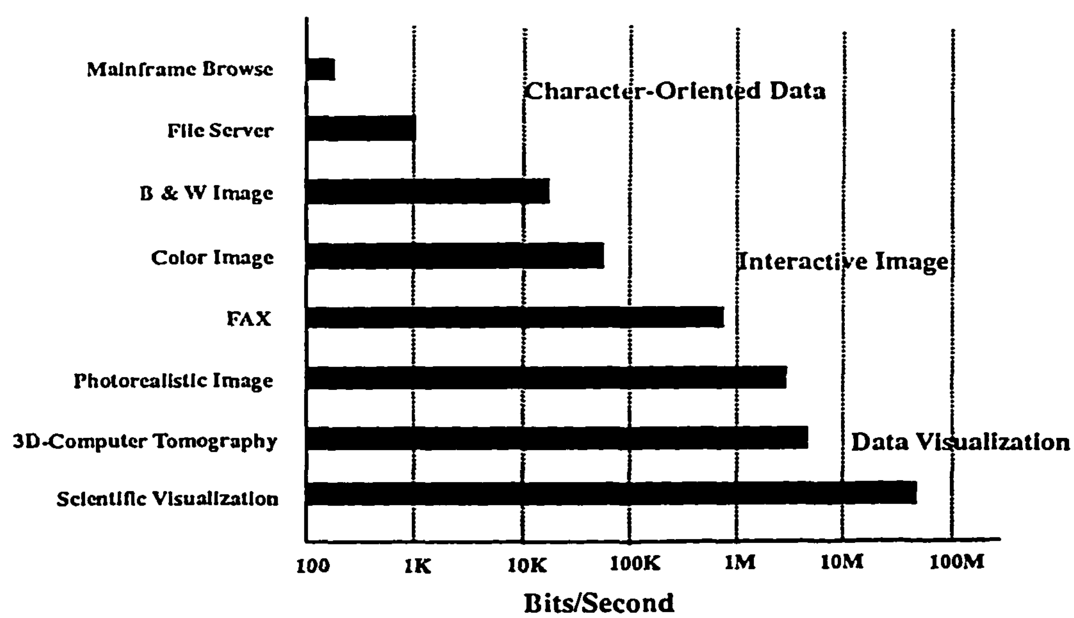


Figure 2.1 : Application Bandwidth Requirements

Data files are quite another matter from transactions. The usual procedure is to break up the files into frames or packets of a size that can be handled by the network. The frames are transferred along with a header. In the header is information about the sequencing and the nature of the file from which they are derived. This information is used by the receiving station to reconstruct the file.

2.1.2 Distributed Multimedia Computing

Multimedia combines the interactivity of a computer with a natural user interface that includes audio, video and real images (Figure 2.2). Over the past decade, multimedia has proven its worth as a uniquely powerful set of technologies when used to support applications in the fields of education, research, business, medicine and entertainment. Most of these applications have been run on standalone systems. For multimedia to reach its fullest potential, it must move from the standalone single user environment to the multi-user distributed environment.

Today two goals can be achieved with off-the shelf products:

- High-end personal desktop computers or workstations with multimedia capabilities can be built;
- Workgroup systems consisting of workstations interconnected via local and wide area networks to a host or server system can be implemented.

In order to construct an open distributed system for multimedia applications, the above mentioned subsystems need to be combined i.e. build workgroup systems where multimedia enabled workstations are interconnected via networks to multimedia servers. The networks must have sufficient bandwidth to carry both computer data and real-time media such as audio and video.

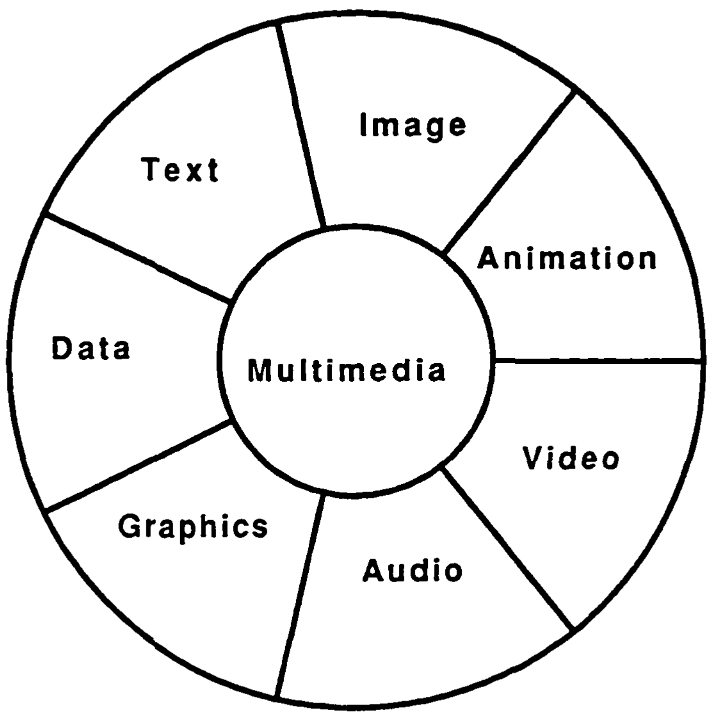


Figure 2.2 : Components of Multimedia

Video and audio files, also known as multimedia objects, are large in size, hence their transmission as continuous data streams when played or captured, and the need to control and synchronize them require key elements of distributed computing to be enhanced to support workgroup and distributed systems [41]. An important objective of a distributed multimedia system is to make the network of heterogeneous systems to appear as a single system to the user. All computing resources and services that a user is authorized to access, local or remote, appear as local resources and services. Accomplishing this transparently can be done using existing transport networking facilities coupled with careful network configuration design. However, to ensure a very robust quality of service, a new capability, network data stream handling which includes some amount of bandwidth management must be included in the open distributed environment. Thus it must be noted that while some applications require real time video transmission facilitated by scalable video, others require only conventional store-and-forward transmission of multimedia data objects.

It must be recognized that in a distributed system the client/server multimedia applications must utilize the network efficiently while maintaining an acceptable level of service [22]. This can be accomplished by a digital video server that uses a scalable digital video and audio compression algorithm to adaptively control video data stream parameters such as frame rate based on feedback about transport network performance. The client system receiving the video and audio data stream decompresses it and presents the video and audio at a quality commensurate to what could be handled by the network without seriously degrading other network users [26, 27]. The encoding method for video and audio data influences quality of service (QoS) parameters, particularly for video.

This dissertation looks at how LAN performance can be reserved to maintain acceptable quality of service to maximum number of concurrent users.

2.1.3 Digitization of Video and Audio

Conventional video signals are the electrical analog of scene brightness. They vary continuously and are described as having an analog format. At the receiver, the eye responds to analog displays with continuous brightness variations. Video signals, therefore, are inherently analog, both when generated and when displayed. Between these endpoints, however, there is the option of converting the signal to digital format.

The analog video signal must be converted to the digital format before it is used in the distributed multimedia environment [28]. The process of converting a video signal to a digital bitstream is called analog-to-digital conversion (A/D conversion), or digitizing. A/D conversion occurs in two steps:

- Sampling - the analog signal is sampled in the time domain at uniformly spaced sampling points; and
- Quantization - the amplitudes of the samples are defined by discrete, pre-established quantized levels, which are identified for each sampling point.

Each level is then described by encoded bits in a byte. The choice of number of bits per representation determines the number of levels that can be identified. If there are too few bits, the picture will have a blotchy appearance. On the other hand, the choice of too many bits unnecessarily increases the storage and bandwidth requirement in the distributed system.

A commonly used format for composite signals employs a sampling rate that is 4 times the sub-carrier frequency, or 14.4 Mhz (NTSC). With an 8-bit word length, the bit rate is 115.2 Mb/s while for an HDTV signal it is 600 Mb/s. The bandwidth required is approximately one-half, i.e 58 MHz and 300 MHz respectively. However, given that the sampled data exhibits a great deal of redundancy and that the video signals possess repetitive and predictive characteristics, compression is applied, thus significantly reducing the amount of bandwidth required for transmission of the stream [14].

The digitized video signal is represented in values that can be represented on display devices as pixels. Each pixel is a point on the display and has a value for intensity and color as part of the static image or video frame. A video sequence is displayed as a series of frames which possesses a definite value for each pixel in the display. When the frames are played back in sequence on the display device, a rendering of the original video data is created. In real-time video the playback rate is 30 frames per second. This is considered as the minimum rate necessary for the human eye to successfully blend each video frame together into a continuous, smoothly moving image. Playback rates of 15 frames per second or higher are acceptable in some instances depending on the amount of motion of the objects in the original video.

There are many encoding methods available that will compress video data. The majority of these methods involve the use of a Fourier or Discrete

Cosine Transform (DCT) coding scheme as shown in Figure 2.3. These transforms physically reduce the size of the video data by selectively discarding unnecessary parts of the digitized information. Transform compression schemes usually drop 10-25 percent or more of the original video data. This amount depends on the content of the video data and upon what image quality is acceptable. Thus, depending on the bandwidth of the original analog signal, the sampling rate, the quantization step, the encoding method and the desired image quality, the resulting data rate for a digital video signal can range from 64 Kb/s to tens of Mb/s.

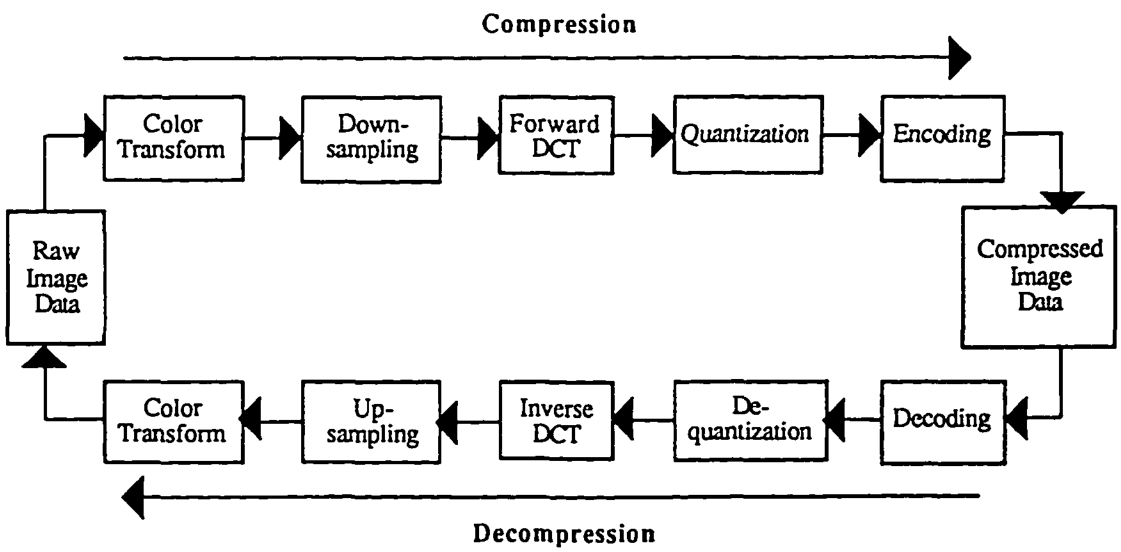


Figure 2.3 : Video and Audio Compression Process

Voice or audio has also been carried as analog signals for quite some time. With the advent of digital transmission media for voice networks and with recorded voice in wide use, voice is now more often digitized and carried as packet files with several voice samples per packet. Storing audio as digital samples is known as Pulse Code Modulation (PCM). PCM is a simple quantizing or digitizing algorithm, which linearly converts all analog signals to digital samples. Differential Pulse Code Modulation (DPCM) is an audio encoding scheme that quantizes the difference between samples rather than the samples themselves. Fewer bits may be used to encode the same sound as the differences are easily represented by values smaller than the values required for the samples. One other audio compression scheme, which uses difference quantization, is Adaptive Differential Pulse Code Modulation (ADPCM). DPCM is a non-adaptive algorithm as it does not change the way it encodes data based on the content of the data it is encoding. DPCM uses the sample number of bits to represent every signal level. ADPCM, specifically adapts by using fewer bits to represent lower-level signals than it does to represent higher-level signals. Many of the most commonly used audio compression schemes are based on ADPCM.

The quality of the audio data is determined by three parameters:

• sample resolution - number of bits per sample which typically is 8 or 16 bits;

• sampling rate - number of times the analog waveform is read to collect data per second which typically has values of 44.100 kHz (high quality), 22.254 kHz (medium quality) and 11.025 kHz (lower quality); and

• number of audio channels sampled which may be one (mono) or two (stereo).

The amount of data produced by sampling even a few seconds of audio is quite large. But audio data contains a fair amount of redundancy that can be removed using algorithms similar to those used for image or video compression.

Audio may be compressed to even lower bit rates, which are needed if it is to be transmitted over narrow band facilities such as radio channels. Since the data rates are fairly low, bandwidth is rarely an issue with digitized audio. Far more often, the problem is the real time transport required to keep conversations going or preventing the breakup of the audio signal. Delays in the tens of milliseconds require echo-canceling electronics, and longer delays distract those trying to talk or listen.

The coding schemes that are implemented can be classified into a hierarchy:

• intraframe compression where each frame is compressed and coded independently. Such coding methods allow QoS variations by decreasing the frame rate through frame dropping. Various dithering algorithms can also decrease the original encoded quality;

• intra- and interframe compression like the MPEG standard described later in this section; and

• layered compression has scalable coding schemes. The video is encoded in different layers, where the lowest layer contains basic information such as luminance, while higher layers carry additional information such as chrominance or increased bits for increased resolution. This scheme allows for optimization of the quantity of data that needs to be transmitted for multimedia delivery in the system.

One of the common video and audio encoding and compression method is the MPEG standard. As mentioned above, MPEG uses two types of compression methods to encode video data, namely, intraframe and interframe encoding [23]. Interframe coding is based upon both predictive and interpolative coding techniques. When capturing frames at a rapid rate, redundant data will be found in any two or more adjacent frames. MPEG method of compression makes use of this "temporal redundancy" and hence does not encode the entire frame of data, as is done in intraframe encoding. Instead, only the differences (deltas) in information between the frames is encoded. This results in greater compression ratios, with far less encoding of data. This type of interframe encoding is called predictive encoding. A further reduction in data size may be achieved by the use of bi-directional prediction. Differential predictive encoding codes only the differences between the current frame and the previous frame. Bi-directional prediction encodes the current based on the differences between the current, previous and next frame of the video data. This type of interframe encoding is called motion-compensated interpolative encoding.

MPEG encodes video streams to support interframe and intraframe encoding with three types of coded frames (Figure 2.4): a) I-frame (intraframe encoded) which contains a single frame of video data that does

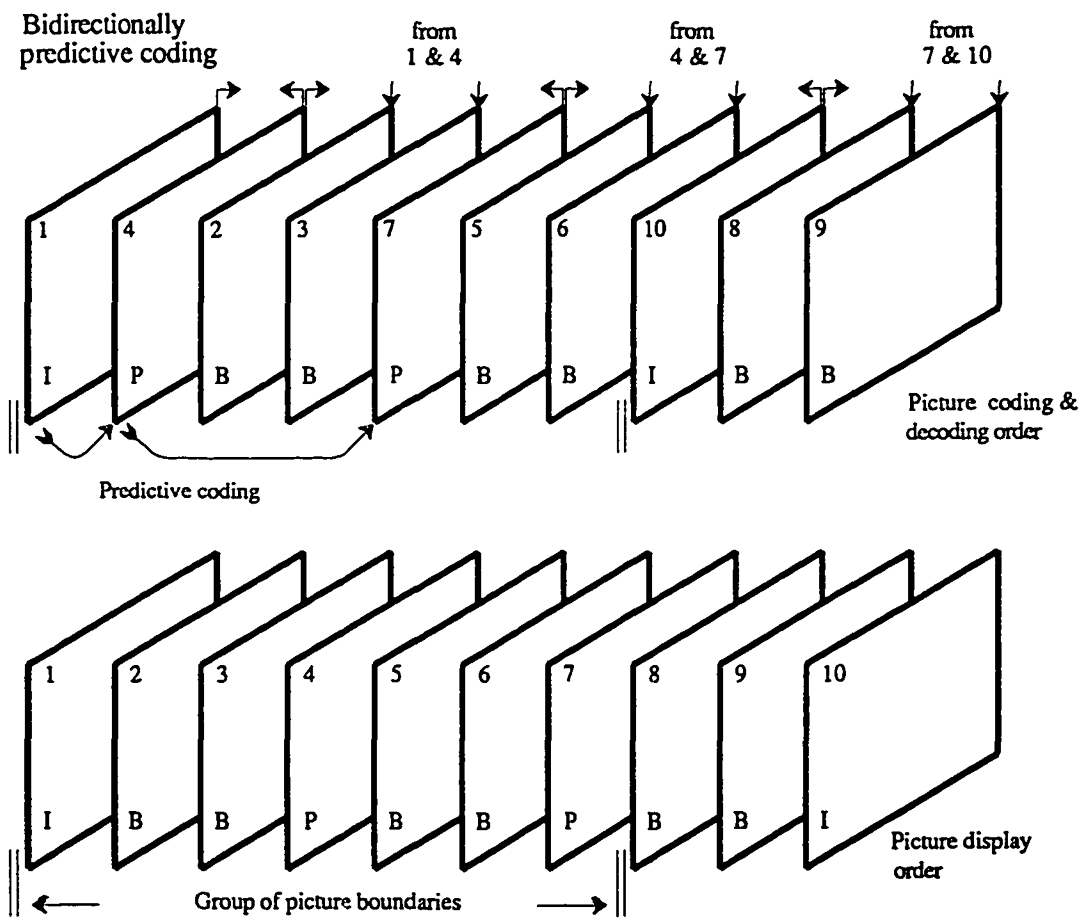


Figure 2.4 : MPEG Coding Loop

not rely on the information in any other frame to be encoded or decoded. Each MPEG data stream starts with an I-frame; b) P-frame (predictive encoded) is constructed by predicting the difference between the current frame and closest preceding I- or P-frame; and c) B-frame (bi-directional encoded) is constructed from the two closest I- or P-frames. The B-frame must be positioned between these I- or P-frames. Once an I-, P-, or B-frame is constructed, it is compressed using DCT compression method to reduce the spatial redundancy and interframe encoding for temporal redundancy.

2.1.4 OSI Model and Multimedia Communication

The process of adding and removing communications protocols to transmitted frames is probably best described with reference to the well-known seven layer Open Systems Interconnection (OSI) model of the International Standards Organization (ISO), which is shown in Figure 2.5.

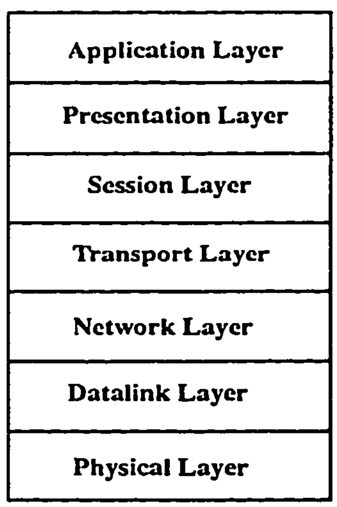


Figure 2.5 : OSI Reference Model

Application Layer: Provides the service elements to process the exchanged information; services include resource sharing, file transfer, and database management. In the case of multimedia, special services for support of real time access and transmission of audio and video must be provided [42].

Presentation Layer: Provides services necessary to format exchanged data and manage session dialog i.e. the layer provides services for transformation between application-specific formats and the agreed-upon format. The multitude of audio and video formats necessitate conversion between formats for transmission and storage on various devices such as CDROM, tape and streaming raid disks which store continuous data.

Session Layer: Establishes and terminates connections; arbitrates user rights to services; synchronizes data transfer. The session layer guarantees the existence of a connection during the session. Types of sessions include point-to-point sessions, connection to many destinations or multicast session and connection to many sources or multidrop session. In the case of continuous media, multimedia sessions which reside over one or more transport connections, must be established. This introduces a more complex view on connection reconstruction in the case of transport problems.

Transport Layer: Provides functions for error-free delivery of messages such as flow control, error recovery and acknowledgement. This layer provides a process-to-process connection. At this layer, quality of service (QoS) is maintained by bridging the gap between what the network layer can provide and the needs of the transport users. Large packets are segmented at this layer and reassembled into their original size at the receiver. Error handling is based on process-to-process communication. When dealing with continuous data streams, the QoS parameters must match their requirements. The error handling scheme does not have the option of retransmission, as this will increase the end-to-end delays and strong jitter. Synchronization between two logical data units (LDU) of one connection and between session data units (SDU) of different connections, is provided by this layer.

Network Layer: Provides transparent routing of messages between two transport entities. The network layer transports information blocks in the form of packets from one station to another across one or more segments of the network or across a number of networks. This layer provides services such as addressing, internetworking, error handling, network management with congestion control and sequencing of packets. In the case of continuous media, this layer must incorporate the features of resource reservation and guarantees for transmission. QoS parameters identifying the requirements for continuous data stream transmissions, define the request for resource reservation. The resources must be reserved along the path between the communicating stations. This process leads to a connection-oriented behavior with correct packet ordering and reduced end-to-end delay with small jitter. If internetworking is included, for different communication structures in multicasting or broadcasting connections, duplication of packets can follow, which may introduce further complexity in the reservation process. The network QoS for a connection should be negotiated at this layer [47].

Datalink Layer: Provides rules for transmission on the physical medium such as packet formats, access rights, error detection and correction, flow control and block synchronization. Access protocols are very much dependent on the network. Networks can be divided into two categories: a) networks using point-to-point connections; and b) networks using broadcast channels, sometimes called multi-access channels or random access channels. In a broadcast network, the key issue is how to determine, in the case of competition, who gets access to the channel. This problem was solved by the introduction of the Medium Access Control (MAC) sublayer and the MAC protocols, such as the timed token rotation protocol and the carrier sense multiple access with collision detection. The MAC sublayer is especially important in local area networks (LANs), nearly all of which use multi-access channels as the basis of their communication. As mentioned before, continuous data streams require reservation and throughput guarantees over a line. If delays are to be kept at a minimum, the error control for multimedia transmission as compared to discrete data transmission, needs a different mechanism than retransmission because a late frame is a lost frame. However, because of the new high-speed networks based on fiber optics, there may not be a need for any error control at this layer. These networks favor multimedia transmission because of their very low transmission error rate. Further, a fixed-size information block (cell) allows for an efficient protocol implementation providing reservations and guaranteed throughput.

Physical Layer: Provides mechanical and electrical level interconnection for stations. For example, the type of modulation and bit-synchronization are important in the transfer of data. With respect to the particular modulation, delays during the data transmission arise due to the propagation speed of the implemented transmission medium and electrical circuits. They determine the maximum possible bandwidth of this communication channel. In the case video and audio data, the delays must be minimized and a relatively high bandwidth should be supported by the transmission medium.

In preparing outgoing packets for transmission, each layer adds its own header to the block of data it receives from above, treating all received data as an inviolate data unit. On reception, each layer examines and acts on its corresponding header, and it passes the rest of the packet to the next higher layer.

2.1.5 Network Technologies

The technologies available for consideration in the design of the network for a distributed multimedia environment can be classified into legacy LANs and High Speed LANs [37]. The legacy LANs consist of the traditional shared media ethernet, token ring and FDDI networks while the high speed LANs consist of switched ethernet and asynchronous transfer mode (ATM). Let us consider in detail these technologies.

2.1.5.1 Legacy LANs

Ethernet is the most frequently deployed LAN technology. Its 10 Mbps bandwidth allows theoretically up to four compressed video streams. The drawback of the Carrier Sense Multiple Access with Collision Detection (CSMA-CD) method is its non-deterministic behavior. In high load situations it exercises no control over access delay or available bandwidth per application. If a traditional application, such as a remote file access, tries to use a large percentage of available bandwidth, no mechanisms exist to assure a fair distribution of bandwidth. Further, ethernet does not provide any access-priority mechanisms and thus cannot give preferred treatment to real-time traffic over conventional data.

The lack of delay guarantees makes ethernet not suitable for distributed multimedia. Nevertheless, many of today's experimental multimedia applications use ethernet as their transport mechanism, generally in a controlled and protected environment. It provides enough bandwidth for a few streams plus a multicast function, which makes it suitable for a small scale distributed multimedia system.

A variant of ethernet, called Isochronous Ethernet has been developed to have the same packet delivery of ethernet together with ISDN-like isochronous channels on a single wire to a terminal. The channels are basically data channels of an ISDN system. This approach has limited bandwidth and lacks multicasting support. It provides truly isochronous traffic, i.e. data with optimal delay characteristics.

The ethernet bandwidth can be scalable to 100 Mbps using the same CSMA-CD access protocol. This Fast Ethernet system has the same limitations with regard to access delay characteristics, since any station on the network can disrupt the multimedia stream through heavy traffic. Multicasting is available in the fast ethernet networks. It is a possible choice for small to medium configuration but is not a good means for multimedia communication.

The Token Ring access protocol is much better suited than Ethernet to support multimedia data. It has the advantages of higher bandwidth i.e. 16 Mbps and the availability of MAC-level priorities. The priorities can differentiate between real-time data (high priority) and normal data (low priority). Control and access to low-priority bandwidth is the same as for ethernet. Stations can apply to a bandwidth manager for a portion of reserved (high priority) bandwidth. The bandwidth manager keeps track of the total allocated bandwidth to prevent over commitment. The scheme works by providing access control or traffic shaping mechanism for each multimedia station to assure that a station does not exceed its allocated bandwidth. Avoiding bandwidth over commitment of token ring does not guarantee acceptable access delays. In addition to the bandwidth management scheme, token ring has limited multicasting capability which makes it a viable entry network for multimedia communication.

The Fiber Distributed Data Interface (FDDI) is conceptually a superset of a fast Token Ring. Thus most of the techniques discussed for the Token Ring apply directly to FDDI but with a bandwidth of 100 Mbps. The larger bandwidth will support a large number of multimedia stations. In addition to priority traffic, FDDI also supports a synchronous traffic class, which allows traffic with bounded delay to use the ring. The delay limit is configurable at the time of ring utilization. This class for synchronous traffic is not widely deployed and hence a bandwidth management scheme is necessary for multimedia communication. Multicast communication is supported by FDDI.

2.1.5.2 High Speed Networks

The use of Asynchronous Transfer Mode (ATM) has gained importance in the past few years with its promise of being able to transport multimedia objects or data [21, 31]. ATM is also known as Asynchronous Time Division Multiplexing (ATDM). Bell Labs have been working in the area since 1969 producing three experimental ATDM LANs: Spider, Datakit and Incon [13]. ATM is currently being standardized for use in the wide area Broadband ISDN [38]. This has caused local area ATM to become popular, with the "ATM Forum" working on standards [5]. There are two key principles in ATM technology that make it beneficial for deployment in networks, namely 53 byte fixed-size data transfer units and establishment of virtual circuits.

ATM is a switched, connection-oriented local and wide-area networking technology that allows a theoretically unlimited number of users to have dedicated, high speed connections with each other and with high performance servers. It is defined by a set of interface standards for both wide area and local area networks. These interface standards offer two advantages:

- since interfaces, rather than internal architectures or implementation details, are defined, interoperability between equipment developers can be maintained and developers are given the flexibility to differentiate their products;
- because the same standards serve as basis for both local and wide area network implementations of ATM, seamless integration of the LAN and the WAN can be achieved using ATM.

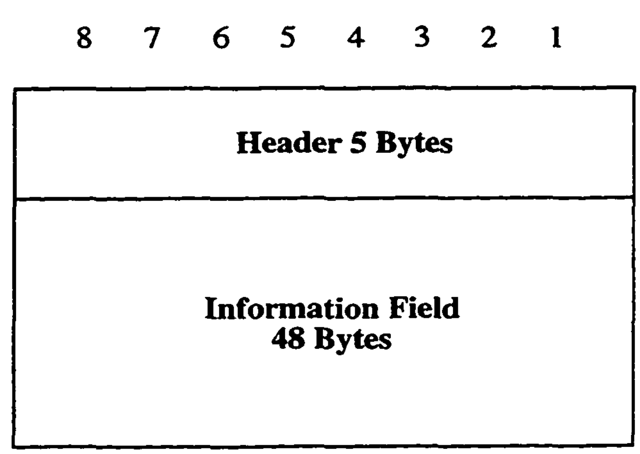


Figure 2.6 : ATM Cell

ATM's primary differences with shared media networks such as Ethernet, Token Ring and FDDI is its use of dedicated media, its connection oriented nature and its fixed length cells [24].

ATM uses dedicated media connections in parallel. Legacy or shared media networks are inherently serial. Users on an ethernet network, for example must contend for one data pipe, wait for an opening, and then send

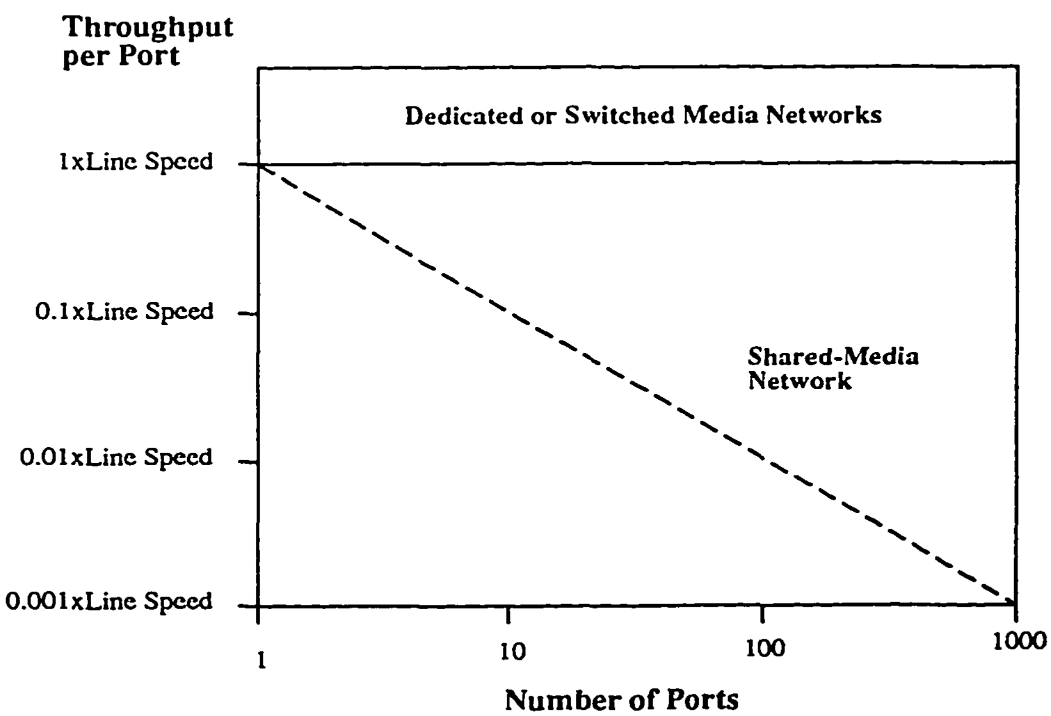


Figure 2.7 : Throughput per port for Shared and Dedicated Network

their transmissions. This means that only one network transmission can take place at a time. The result of the serial nature of shared media LANs is that the average bandwidth available to each user declines with each additional user. If one user is using 10 Mb/s ethernet network, that user's average bandwidth is 10 Mb/s. If 10 users are on the same ethernet segment, however, that average drops to 1 Mb/s, and so on as seen in Figure 2.7.

Switched Ethernet and ATM supply each user with a dedicated media connection to a switch. The switches permit many of these dedicated media connections to run in parallel. This means that multiple conversations may take place simultaneously through a single switch. These switches may, in turn, be connected together to form networks. Again many connections can run in parallel, allowing multiple conversations to take place over multiple paths through-out the network at the same time. Thus the arithmetic of decreasing average available bandwidth as the number of workstations increases, as in shared media LANs, is not applicable in an ATM or switched ethernet network. The maximum bandwidth and the average bandwidth per connection are the same: 155 Mb/s in the case of today's ATM, scalable to 622 Mb/s and beyond in the future.

The second difference between shared media networks and ATM is that ATM is connection-oriented or the concept of virtual circuits. In a connectionless shared-media LAN, the header of each packet must include sufficient information to route a packet to its destination. Before data transfer can occur between two points on an ATM network, however, a connection needs to be established between the source and the sink using a signaling protocol. Once this route is setup the ATM cells are self-routing, as each cell contains information identifying the established connection or circuit for its use. Therefore, there is no need to repeat complex routing information in each cell. This idea of circuits is analogous to the circuits used in the telephone system i.e. a dedicated link is made between the end-points for the duration of the communication. The virtualization of this idea is to multiplex many communication channels onto a single physical line as shown in Figure 2.8.

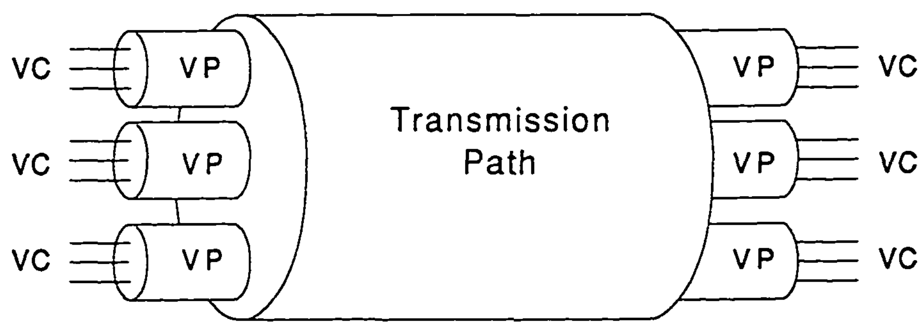


Figure 2.8 : ATM Virtual Path(VP)/Virtual Channel(VC) structure

In shared media networks, end nodes take the variable length packets transferred to them by upper layer protocols, append trailers and headers, and send the data on the network in the form of variable length frames. In an ATM network, end nodes take the variable length packets transferred to them by upper layer protocols and map these packets into fixed length cells with a five-byte header followed by a 48-byte payload.

Specifically in the five-byte header shown in Figure 2.9, each ATM cell contains two fields - a Virtual Path Identifier (VPI) and a Virtual Circuit

Identifier (VCI), which jointly serve to identify connections. The VPI/VCI have local significance only i.e. they identify a particular cell to be associated with a particular virtual circuit, across a single link. A set of VPI/VCI mappings is programmed across an ATM network from one terminal to the other (and vice versa), through any number of intermediate switches, that identifies cells belonging to the connection on each link. It must be noted that there is no reservation of specific positions in the frame for each circuit, unlike in other networks. ATM can reserve bandwidth over a period of time, but it does not dictate the position in the stream of data. This means that bandwidth is used only when a node has data to send, which allows ATM to incorporate the efficiency of packet switching, namely the ability to allocate as per the needs and the demand.

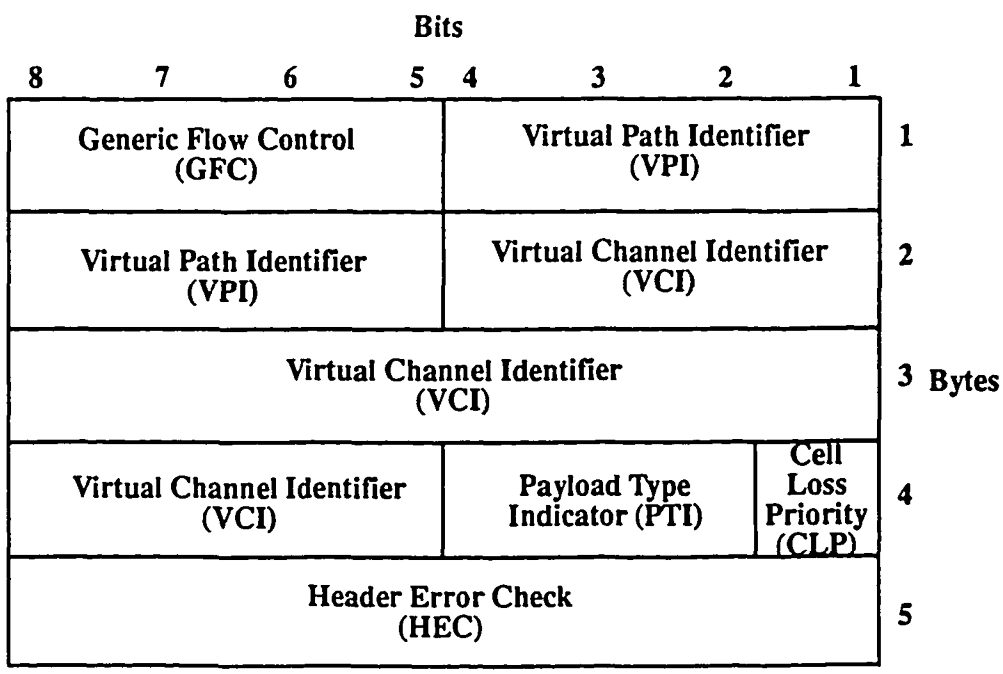


Figure 2.9 : ATM Header Structure

The ATM protocol reference model shown in Figure 2.10, is similar to the OSI layered model. Communication from higher layers occurs through three layers -the Physical Layer, the ATM Layer and the ATM Adaptation Layer (AAL). The model differs from the OSI model, however, in its use of "planes" which are analogous to protocol suites. The portion of the layered architecture used for end-to-end or user-to-user data transfer is known as the User Plane (U-Plane). Similarly, higher layer protocols are defined across the ATM layers to support signaling - this is the Control Plane (C-Plane).

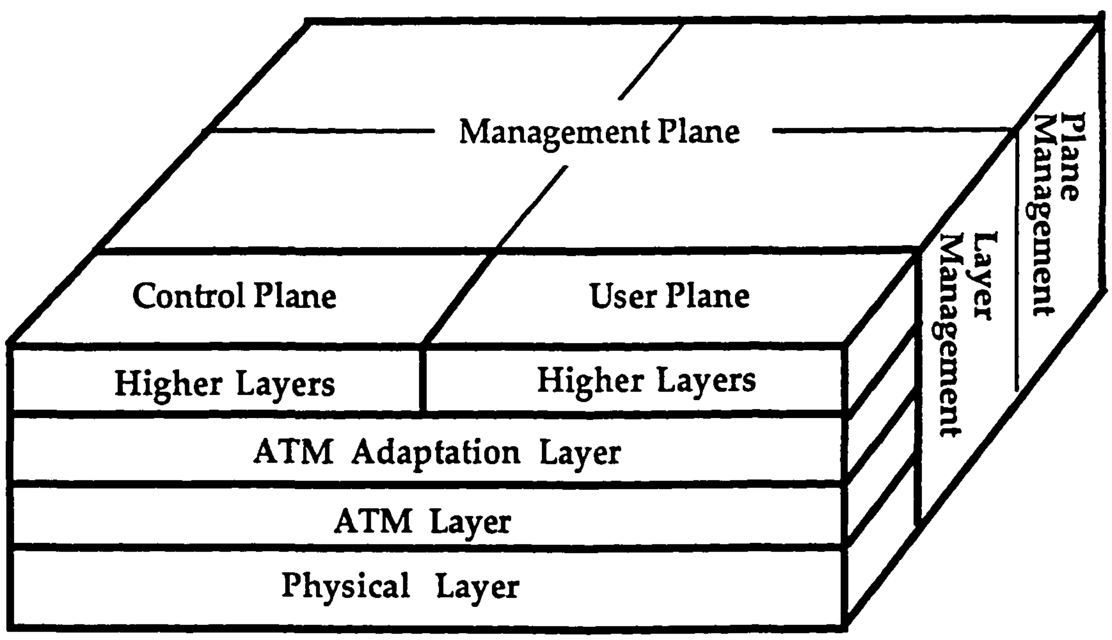


Figure 2.10 : ATM Protocol Reference Model

A management plane provides the control of an ATM node. The Management Plane (M-Plane) is further subdivided:

1) Layer Management is used to manage each of the ATM layers, with a management entity corresponding to each ATM Layer.

2) Plane Management is concerned with the management of all other planes.

In practice, the ATM protocols can be understood simply in terms of the function of each of the ATM protocol layers, as with conventional LAN protocol stacks.

The physical layer is segmented into two sublayers: the Physical Medium Dependent (PMD) Sublayer, and the Transmission Convergence (TC) sublayer. This sublayering is important since it separates transmission from the physical interface, and allows ATM interfaces to be built on a large variety of physical interfaces. The PMD is specific to a particular type of physical layer and deals with factors such as bit timing and the physical medium. The TC sublayer performs a convergence function which receives a bit stream from the PMD and extracts cells to pass to the ATM layer. Some of the common functions of the TC are: cell delineation, cell rate decoupling, HEC generation and checking, and transmission frame generation, adaptation and recovery.

The ATM layer is fully independent of the physical medium used to transport the ATM cells and thus of the physical layer. The following functions are performed by this layer:

- The multiplexing and demultiplexing of cells of different connections (identified by different VCI and/or VPI values) into a single cell stream on a physical layer.
- A translation of the cell identifier, which is required in most cases when switching a cell from one physical link to another, in an ATM switch or cross connect. This translation can be performed either on the VCI or VPI separately, or on both simultaneously.
- Providing the user of a VCC or VPC with one quality of service (QoS) class, out of a number of classes supported by the network. Some services may require a certain QoS for one part of the cell flow of a connection, with a lower QoS for the remainder. The distinction within the connection is made by means of the CLP bit in the cell header.
- Management functions: the header of user information cells provides for a congestion indication and an ATM user to ATM user indication. Pre-assigned VCI values are defined for F4 segment associated and end-to-end associated flows, and dedicated PTI codes for F5 segment associated, end-to-end associated flows and resource management cells. When the PTI does not indicate user information, further information concerning the type of layer management will be found in the information field of the cell.
- Extraction (addition) of the cell header before (after) the cell is being delivered to (from) the adaptation layer.

- Implementation of a flow control mechanism on the user-network interface. This is supported by the GFC bits in the header.

The ATM Adaptation Layer enhances the service provided by the ATM layer to a level required by the next higher layer. It performs functions for the User, Control and Management planes, and supports mapping between the ATM layer and the next higher layer. The functions performed in the AAL depend on the higher layer requirements.

The adaptation layer is divided into 2 sublayers: the segmentation and reassembly sublayer (SAR) and the convergence sublayer (CS). The main purpose of the SAR sublayer is segmentation of the higher layer information into a size suitable for the payload of the consecutive ATM cells of a virtual connection, and the inverse operation, reassembly of contents of the cells of a virtual connection, into data units to be delivered to the higher layer. The convergence sublayer performs functions like message identification, time/clock recovery, etc.

The CCITT has defined four classes of ATM services as shown in Figure 2.11:

Class A: A time relation exists between source and destination. The bit rate is continuous (constant) as in applications such as pulse code modulation (PCM) telephony.

Class B: A time relation exists between source and destination for a connection-oriented service. This includes variable - bit - rate non-data application such as compressed video.

Class C: A connection-oriented data application which does not require a time relation between source and destination and the bit rate is variable.

Class D: A connectionless data application which also does not require a time relation between source and destination.

	Class A	Class B	Class C	Class D
Timing between Source and Destination	Required		Not Required	
Bit Rate	Constant	Variable		
Connection Mode	Connection Oriented			Connectionless

Figure 2.11: ATM Service Classes for Adaptation

To handle these different classes of service, five ATM adaptation layers have been defined by CCITT and the ATM Forum.

AAL - 1: Defined to support class 1 applications

AAL - 2: Defined to support class 2 applications

AAL - 3/4: Layers 3 and 4 were combined to support connection-oriented data service and connectionless datagrams

AAl - 5: As a result of the complexity with AAL - 3/4, several companies proposed AAL - 5. It provides more limited functions (error detection but not recovery) but has lower processing and bandwidth requirements.

Existing LAN technologies can be accommodated in the ATM world by using LAN emulation [19]. In essence, LAN emulation is a way to render the ATM switching fabric invisible to legacy LANs. LAN emulation also enables applications on legacy LANs to access ATM-attached servers, workstations, routers and other network equipment. LAN emulation forwards upper-layer protocols across ATM connections without requiring any modifications to legacy software. At the same time, it makes it possible to convert LAN packets into ATM cells without generating too much overhead at the ATM-attached devices.

An emulated LAN has two main components: LAN emulation clients (LECs) and a single LAN emulation service. LEC software can be deployed in the converter or as part of the driver in an ATM-attached network server or other device. The client software has several jobs. One of the most important being to map MAC (media access control) addresses to ATM addresses which is also known as address resolution. The software that delivers the LAN emulation service is implemented on three logical servers - the configuration server, the LAN emulation server (LES), and the broadcast and unknown server (BUS). Working together, the servers on an emulated LAN perform three functions: transfer data point-to-point between one end-station and another (unicast); transfer data point-to-multipoint from one end-station to many (broadcast or multicast); and resolve MAC addresses to ATM addresses.

The LES is the command and control center for an emulated LAN. It's responsible for registering and resolving MAC addresses or route descriptors to ATM addresses. The BUS takes care of all the broadcasting and multicasting performed over the ATM network. The configuration server furnishes configuration information about the ATM network. It also supplies the address of the LES to the client. LECs communicate with theses servers via two types of ATM connections. Control connections are used for housekeeping tasks such as finding the address of another client. Data connections are used for everything else. The key control connection runs between the client and the LES and is set up by a client when it joins an emulated LAN. This is a bidirectional point-to-point link.

Thus, overall, LAN emulation is the means for combining legacy LAN and ATM technologies.

2.2 Related Work

The following research projects were found in the literature to have setup similar networks with limited focus or operation parameters. Most of the projects are in the field of medicine and are used for archiving, teaching, videoconferencing, collaboration and consultation during surgical procedures.

• The MedNet project at the University of Pittsburgh [44] is used for real-time monitoring, multiparty consultation and collaboration during brain surgery. Phase I of the system incorporated real-time data streams of digitized audio, extensive physiological and computer generated data. Video streams were transported by a parallel, broadband, analog network to the workstation The data network comprised of a shared Ethernet backbone. In phase II the systems were combined to provide only one network for the delivery of digital streams of video, audio and computer generated data. This project uses only ATM to the workstation and does not consider any legacy LAN topologies such as ethernet or token ring. Mednet uses a server in its operation to archive data and distributed to one user at a time. Guarantees on quality of service at the workstation are the responsibility of the user. Buffering is used to obtain synchronization.

• The BERKOM project in Germany [6] was established to study the requirements of a multimedia communication and collaboration system. The project studied the requirements for a multimedia collaboration service, multimedia mail service and multimedia transport system. This project is mainly used connect sites all over Germany using Vermittelndes Breitband-Netz (VBN) network. The LAN environment was ethernet and connected to the VBN using a special multiplexer. This project uses ethernet or FDDI to the workstation and VBN for the wide area connectivity. The project did not use ATM in the initial research. Future activities will include the deployment of ATM to replace the VBN. BERKOM project is researching the uses of multimedia in collaboration and communication. Guarantees on quality of service are implemented in a specialized protocol XTI. Most of the research concentrates on the network QoS. Research into the application QoS is limited and is achieved mainly by using large buffers.

- The High Definition Distance Learning (HDDL) project [43] at the University of Minnesota has been designed to deliver simultaneous video, audio and dual digital slide projection over ATM connections. This project uses two way communication over the wide area for delivery of lectures and discussions. The HDDL project uses ATM connections to Silicon Graphics workstations. HDDL is basically used to deliver lectures and two way video-conferencing. There are no plans to deliver past lectures using a video or multimedia server. The ATM network is basically used for delivery of the data before the time of delivery. Thus, the network is not used as a realtime delivery system for delivering lectures. Protocols such as FTP are used for transmission of the files.

- Medusa project [53] at Olivetti Research aims to provide a networked multimedia environment in which many streams of multimedia data, perhaps thousands are active simultaneously. Medusa's software model uses active objects called modules to represent cameras, displays, format converters and so on. These modules connect together to form applications. The Medusa project uses ATM but it is used as an internal bus and interconnecting network. The multimedia peripherals are connected to an ATM switch and then to the workstation instead of directly to the station. Medusa has a peer-to-peer architecture rather than the hierarchy of client-server architecture. Application may take a hands-on approach where it interacts with the stream coming from the source to the sink or take a hands-off approach and simply connect source to a sink without ever touching the raw data itself. Quality of service parameters are being programmed into the special interfaces which go directly into the multimedia object or module such as a camera, a microphone, monitor, speakers and the workstation itself. The modules developed under this project are specialized and eventually marketed to specific industries.

- At the University of Cambridge, the Desk Area Network (DAN) research [16] focuses on building a multimedia workstation that employs a single ATM switch for interconnecting multimedia devices. DAN investigates the requirements of a multimedia workstation namely, I/O subsystem requirements, operating system modules, caching, etc. The DAN project takes local area ATM techniques and experience and uses them to build an intra-machine interconnection network. The multimedia peripherals are connected to the ATM bus which is then connected to the workstation. The

DAN project concentrates on the connection of multimedia peripherals to an ATM bus. Quality of service is considered when transferring data on the ATM bus which connects the various peripherals. The requirement is that the QoS requirements at the interface of the workstation is communicated to the bus interface of the peripheral.

- The Pandora project [18, 34] investigated the management of time critical data streams and first generation multimedia applications. "The Pandora Box" was produced to add audio and video capabilities to an existing workstation. An ATM network, the Cambridge Fast Ring, is used to interconnect the boxes. Communication is possible between the Pandora workstations to an assortment of servers. The Pandora project uses only ATM. The ATM network is used to connect the various workstations with each other. Legacy LAN topologies are not considered in the project. Pandora network has very limited use of video servers as the network is mainly used for video conferencing. Quality of service parameters are programmed into the special interfaces which go directly into the multimedia object or module such as a camera, a microphone, monitor, speakers and the workstation itself.
- VuNet [1] is a gigabit-per-second local area ATM network implemented at the Massachusetts Institute of Technology. It is implemented in the context of ViewStation. VuNet consists of a set of ATM switches and links. The ATM switches are used to interconnect workstations and multimedia

peripherals, such as video capture boards, image processing systems and audio capture boards. The workstations are general off-the-shelf systems with a VuNet host interface card. This project uses only ATM to the workstation. Video and audio stream are transmitted from the digitizing station to multiple delivery stations. The project concentrates on the delivery of live multimedia streams for videoconferencing applications. Guarantees on quality of service at the workstation are implemented in the VuNet host adapter card and the ATM switch. This project focuses on getting real-time data such as voice and video from the network and delivering it to the application.

Summarizing all the related work and making comparisons with our project we obtain Table 2.1. Our project implements a network which looks at both legacy LAN technologies and ATM technology. We also look at the distributed multimedia environment in detail which includes server architecture and network performance for multimedia applications.

Table 2.1 : Summary of Objectives for Related Research

Name of Project	Use of ATM	Use of Legacy LANs	Use of Multi-Media or Video Servers	Used for Video Conferencing and Collaboration	Quality of Service Implemented Using this Mechanism
MedNet	Yes	No	No (Single User System)	No	Limited to User controls
BERKOM	Yes in the Future	Yes FDDI and Ethernet	No	Primary use	Uses a special protocol to establish network QOS
HDDL	Yes	No	No	Primary use	Network is used to perform staging of data for lectures. No QOS for Video Conferences
Medusa	Yes as Network and W/s Bus	No	No servers but connections directly from source to sink	Primary Use	Uses the special interfaces designed for attaching video sources and sinks. QOS built into the interface

(Table continued)

Name of Project	Use of ATM	Use of Legacy LANs	Use of Multi-Media or Video Servers	Used for Video Conferencing and Collaboration	Quality of Service Implemented Using this Mechanism
DAN	Yes as W/S bus	No	No	Primary Use	Using ATM Bus within the workstation
Pandora	Yes	No	No	Primary Use	Uses Special interfaces
VuNet	Yes	No	No	Primary Use	Uses VuNet adapter
Our Project	Yes	Yes	Yes	Yes	Best Effort but guaranteed with new framework

Chapter 3

Design Specification For An Integrated Network

3.1 Guidelines

When designing any network it is important to consider the demands that will be placed on the system by the different types of data. The data with the most complex demand characteristic will determine the structure of the network. This is due to the fact that this data will put maximum strain on the system. It is assumed all other types of data have requirements that are less stringent than the most complex needs. In our environment the multimedia streams will place the most strain on the system. Traditional transactional data communication needs will be met if we design for multimedia data streams.

3.2 Multimedia Communication Requirements

Multimedia communication requirements impact various aspects of the integrated network design. As seen in the previous chapter the key parameters that must be monitored in this network are bandwidth, transmission delay, capability of multipoint communication, reliability and channel synchronization. Let us consider each of these in further detail.

- Bandwidth - DVI and MPEG standards for video and audio compression suggest that for good quality the data rate should not be lower than 1.5 Mbps.

With regard to wide-area transmission cost, existing H.261 implementations show that 64 Kbps is only acceptable in almost static "talking head" video signals, whereas the 384 Kbps (six bundled ISDN channels) variant yields acceptable results even in a more general environment. This bandwidth is required for simplex, i.e. unidirectional, streams because multimedia traffic is highly asymmetrical. Thus for our design a minimum of 1.5 Mbps must be guaranteed in each direction to each workstation.

- Transmission Delay - International Communications Union (ITU) standards suggests a maximum total end-to-end delay of up to 150 ms for interactive video applications. Based on this delay parameter, traffic may be divided into the following categories: (i) asynchronous type that has no restriction on the transmission delay; (ii) synchronous type that has an upper bound for the transmission delay for each message; and (iii) isochronous type that has an upper bound for the maximum transmission delay and also requires a constant transmission delay for the different packets. Video and Audio communication falls into this last class.

Isochrony does not have to be maintained across the entire path from source to sink but only at the information's final destination. Thus isochrony can be recovered by the use of buffers at the destination. The recovering process will introduce some delay. The end-to-end delay can be broken down into at least four contributing pieces: (i) source compression and packetization delay; (ii) transmission delay; (iii) end-system queuing and synchronization (playout) delay; and (iv) sink decompression, depacketization and output delay. Video streams require the handling of 25 to 30 frames per second. Thus real-time compression or decompression times must not exceed 30 to 40 ms. Using another frame period for queuing and playout delay, leaves 60 ms for the maximum transmission delay.

- Reliability - Traditional communication aims at providing reliable end-to-end communication between two peers. Existing communication systems always use checksum and sequence numbering for error control and some form of negative or missing positive acknowledgement with packet retransmission handshake for error recovery. If check summing is not performed either in hardware at the media access control or link layer, it can affect the system performance drastically. The acknowledgement with subsequent retransmission handshake adds more than a round-trip delay to the transmission of this data. For time critical data such as video and audio streams, the retransmitted packet may be useless. Therefore, in the case of the time sensitive transmissions, the network must let the error control and recovery schemes be handled by the higher communication layers. They can provide the level of reliability required, taking into account the impact on the delay characteristics. A possible remedy for the conflicting goals of reliability and low delay is to use forward error correction (FEC) techniques. The issue of reliability becomes even more complex for multipoint communication.

- Channel Synchronization - When audio, video and other data streams are delivered from different sources via different routes, it is necessary to synchronize these different streams at the sink. Synchronization can be achieved using a combination of time-stamping and playout buffers. In our design we assume synchronization to be built into the operating system on the client and the server.
- Multipoint Communication - Multimedia communication involves audio and video broadcast information. Thus the integrated multimedia communication needs to support multicast communication patterns in addition to normal point-to-point communication.

3.3 Multimedia Network Specification

An infrastructure constructed of the legacy LAN technologies described in the previous section, might be sufficient for supporting multimedia applications in a small scale. A small group of users who retrieve video from a server or participate in a video conference may be serviced by a single ethernet. However, as the number of multimedia-capable users grows, the aggregate bandwidth of the network must also grow to support their communication requirements. Further, videoconferencing applications are not likely to stay restricted to the individual workgroups and hence may make extensive use of the backbone.

Networks are subdivided into subnetworks due to geographical, administrative and addressing reasons. The subnetworks are interconnected by a backbone. Let us consider the traffic generated within a subnetwork is divided into two fractions: the intra-subnet traffic which is uniformly addressed to the other subnetworks on the backbone network, and is proportional to their sizes. In this section a traffic model is presented that makes it possible to evaluate the required bandwidth for the backbone network.

Subnetworks are composed of segments; a segment is a shared channel interconnecting a number of users. Segments may be ethernet, token rings, FDDI and are connected by switches or bridges, which isolate the traffic local to the segment, repeating only the traffic to/from the other segments in the subnetwork. If the bandwidth requirements are low, a subnetwork might be composed of a single segment. However, as bandwidth requirements grow and exceed the capacity of a single segment, two options exist to provide the service: (i) increase the capacity of the segment; and (ii) increase the concurrency by using multiple segments. Option (i) can be exercised up to a certain limit, and is usually not very attractive, because all nodes must be retrofitted with interfaces for higher speed. With option (ii), no modification is needed in the user stations, and the increase in bandwidth is realized by increasing the number of segments, thus allowing multiple concurrent transmissions to take place. Some sort of switching function must be provided between those multiple segments.

The simplest way to implement this switching function is by interconnecting the segments with bridges, but for uniformly addressed traffic, this technique cannot lead to large improvements over the capacity of a single segment. The reason for this lack of improvement in the aggregate throughput is that some of the segments are used to implement the switching function between the concurrent channels and become bottlenecks. If a high capacity channel is provided to carry traffic between segments then substantial bandwidth will be available for inter-segment communication. This channel may be an FDDI ring or a high-speed backplane inside a switching hub. The topology of the subnetwork is a star with a fast and large capacity channel in the center. The evolution of the backbone is similar to the construction of the subnetwork, but on a larger scale.

3.4 Computational Characterization of Network Workload

We present a computational model that will enable us to identify the type of network required to be installed in a distributed multimedia system. Let us consider the following variables:

N^{TOT} : Total number of users on the network

S : Number of subnetworks in the network

N_i^{SUB} : Number of users in subnetwork i, i= 1.....S

G_i : Number of segments in subnetwork i, i=1....S

N_{ij}^{SEG} : Number of users in segment j (j = 1....G_i) of subnetwork i, i= 1.....S

T_{ij}^{SEG} : Traffic being generated in segment j (j = 1....G_i) of subnetwork i (i= 1.....S) in bits/second

T_i^{SUB} : Traffic being generated in subnetwork i (i= 1.....S) in bits/second

T^{BACK} : Traffic in the backbone

B_{TEXT} : Data rate of the regular data or text stream in bits/second

B_{MULT} : Data rate of the multimedia stream in bits/second

γ : Fraction of the traffic generated in the segment destined to users in the same segment

α : Fraction of the traffic generated in the subnetwork destined to users in the same subnetwork

δ : Fraction of the users in the network which are multimedia capable

Multimedia may be introduced gradually and hence only a fraction δ of the total number of users on the network will be capable of generating multimedia traffic. If we assume that each multimedia capable user generates a stream of data rate B_{MULT}, then the traffic generated in a segment by these users is:

$$T_{ij}^{SEG} = \delta\, N_{ij}^{SEG}\, B_{MULT} \tag{1}$$

A fraction γ of T_{ij}^{SEG} remains in the segment, and the remaining (1-γ) is transmitted into the subnetwork i. Thus the traffic in the subnetwork is:

$$T_i^{SUB} = \sum_{j=1}^{G_i} (1-\gamma)\, T_{ij}^{SEG} \tag{2}$$

If we consider the subnetwork consisting of switching ethernet hubs then each port is considered as a segment. Further, in a distributed multimedia system due to the bandwidth requirements of transmitting video and audio streams, one may attach only one workstation per port of the hub. In this case, each segment contains only one user and hence $\gamma = 1$. Equation (2) may be written as:

$$T_i^{SUB} = \delta\, N_i^{SUB}\, B_{MULT} + (1-\delta)\, N_i^{SUB}\, B_{TEXT} \tag{3}$$

This is the traffic generated by the users capable of generating multimedia streams and the rest of the users generating data or text streams. If we consider a completely distributed multimedia system then every station in the subnetwork has the capability of either generating the multimedia stream or at least viewing a multimedia stream. This leads to the variable δ=1. Hence:

$$T_i^{SUB} = N_i^{SUB}\, B_{MULT} \tag{4}$$

If each of the workstations were setup for video-conferencing or to handle multiple streams simultaneously then we can modify equation (4)

$$T_i^{SUB} = \beta\, N_i^{SUB}\, B_{MULT} \tag{5}$$

The maximum number of streams that the workstations can handle is represented by β.

If each segment connects a number of users or workstations and a subnetwork consists of a number of segments, then the fraction of traffic between segment i destined for segment j is represented by:

$$\alpha_{ij}^{int} = \frac{N_j^{SEG}}{N^{SUB}} \tag{6}$$

Some of the traffic ($\alpha\, T_i^{SUB}$) generated in the subnetwork i may be sent to users in the same subnetwork while the remaining $(1\text{-}\alpha)\, T_i^{SUB}$ is transmitted to the backbone. The traffic in the backbone is:

$$T^{BACK} = \sum_{i=1}^{S} (1-\alpha)\, T_i^{SUB} \tag{7}$$

The traffic from subnetwork i to subnetwork j is considered to be proportional to the size of subnetwork j, i.e

$$\alpha_{ij}^{ext} = \begin{cases} \dfrac{(1-\alpha)N_j^{SUB}}{N^{TOT} - N_i^{SUB}} & \text{if } i \neq j \\ \alpha & \text{if } i = j \end{cases} \tag{8}$$

The traffic from subnetwork i into the backbone, T_i^{in}, is given by:

$$T_i^{in} = T_i^{SUB}(1-\alpha) \tag{9}$$

The traffic from the backbone into subnetwork i, T_i^{out}, is given by:

$$T_i^{out} = \sum_{\substack{j=1 \\ j \neq i}}^{S} \frac{(1-\alpha)N_i^{SUB}}{N^{TOT} - N_j^{SUB}} T_j^{SUB} \tag{10}$$

The traffic in the backbone can also be represented by the following equation

$$T^{BACK} = \sum_{i=1}^{S} (1-\alpha)\,\beta\, N_i^{SUB}\, B_{MULT} \tag{11}$$

The backbone traffic is found to be proportional to the following:

- the fraction of traffic generated in the subnetworks that is destined to other subnetworks;
- the number of streams that each workstation can handle simultaneously when attached to the network;
- total number of users on the network; and
- bandwidth of the multimedia streams.

3.5 A Generalization for the Multimedia Server Architecture

Sharing applications with digital video and audio content in a distributed computing environment poses unique challenges to the capabilities of traditional servers [49]. The applications can be divided into two main categories: (a) stored video applications which involve sharing of digital information stored in a server through a local area network, and (b) live video applications which involves the use of the video medium for interactive communication among humans such as video conferencing and collaborative computing.

The characteristics of digital video files and traffic differ substantially from those encountered with traditional data applications: (a) video files are quite large i.e. in some cases a video file is as large as a complete database; (b) video traffic is continuous in nature while data traffic is bursty; and (c) the data rate of a video stream is relatively higher than the mean data rate of a single data traffic source. Thus, the commonly employed data file servers are not suited to support video services over local area networks. New multimedia servers or data servers with additional capabilities are needed to handle the characteristics of digital continuous media files and traffic [29, 50].

In the case of data applications, a user expects a fast response time for a file access request to a server or data transmission request on a network, as compared to the time it takes to place the next request. Hence, the capacity of a server and the overall network bandwidth must both be large as compared to the average demand placed by a single user. Accordingly, file servers designed for supporting data applications and network designs to carry data traffic have been based on the principle of bandwidth sharing and statistical time multiplexing. File servers have furthermore taken advantage of the property of locality in file access and incorporated appropriate caching mechanisms. In all cases, as the overall load placed on the shared resources increased the average response time experienced by all users also increased.

Multimedia servers need to control very large volumes of continuous datastreams, the media devices that store these datastreams, and the networks and communications that distribute the datastreams to and from the users in real time, in order to facilitate shared access to data. There are three basic approaches to the design of servers aimed at supporting multimedia applications. The first approach is to retrofit a data application file server in such a way as to allow it to handle video traffic; for example, a Novell server may be equipped with a Network Loadable Module (NLM) which provides the streaming capability needed for video. While it is expected that with this approach a single server can support both multimedia and data applications simultaneously, there may be a compromise in the performance for both. A second approach is to design a server entirely dedicated to video applications; such a server is then designed and optimized specifically for streaming, and can thus offer the best performance in video service. This video server needs to coexist with the data application server, in order to support both data services and multimedia applications. A third approach is to design a fully integrated server which is capable of both transactional data and streaming video services in a well coordinated and dynamically optimized fashion.

If we consider the second option then the bandwidth capacity requirement for the network connection can be obtained by using the variables defined in the previous section. If β is the number of video streams, each having a data rate of B_{MULT} bits/second, that the video server distributes simultaneously to each multimedia capable workstation on the network, then the capacity of the interface is:

$$B_{SERVER} = \delta\, \beta\, N^{TOT} B_{MULT} \qquad (12)$$

It is assumed that the video server is connected to the backbone and servicing stations on all the subnetworks that are multimedia capable. In the above equation $\delta = 1$ if all the workstations are multimedia capable in a distributed multimedia environment. If the server handles not only store-and-forward multimedia files but also live video streams then all the workstations may be watching the live video broadcast from the server, giving rise to δ=1.

Multimedia also has a significant impact on the storage requirements, due to the size of media objects. For a reference point, a 500 page text book requires 1 MB of storage. Ten fax quality images require 640 KB, whereas ten color or detailed images require 75 MB. Five minutes of uncompressed voice-quality audio requires 2.4 MB of storage and 52.8 MB of storage are required for compact disc quality digital audio. Digitized video requires the greatest storage capacity of all data forms. Without compression techniques, practical storage of digital video is impossible. A two hour television quality video can be compressed to about 2 GB of storage.

Four characteristics must be found in the multimedia storage system:

• there must be enough storage capacity as required by the application;

• it must have the capability of random access for both recording and playback;

• the I/O throughput must be sufficient to support simultaneous access by a fixed number of users; and

• the latency between a request and delivery for new streams must be within a constant limit.

Furthermore, when serving a video stream, it is imperative that continuity of the stream is maintained. In the case of playback, data must be retrieved from the disk(s), transmitted over the network, and made available to the decoder no later than the time at which it is needed so as to avoid letting the decoder underflow. Similarly, when a stream is getting recorded, the writing of data on the disk must keep up with the rate at which it is getting produced so as to avoid buffer overflow and data loss. Thus, to maintain continuity, every I/O operation must be completed within some stringent time constraint.

An effective multimedia server will allocate multimedia objects to the right place at the right time [52]. The allocation is dependent on the storage media, i.e. type and size, available in the server. The storage medium type is characterized by its practical and technological storage capacity, and its bandwidth for storage or retrieval. Multimedia servers usually feature a large storage capacity across different storage media such as a tape library, disk storage and soft storage in the form of RAM modules and cache. A tape library can hold 1 to 15 terabytes of data, but the bandwidth is limited to a few MB per second. A disk system may contain 5 to 100 gigabytes of data and a disk arm can read at a speed of 1 to 5 Mb per second. The storage of RAM or cache is generally up to 32 to 512 MB. The RAM or cache bandwidth is determined by the system bus speed. Thus the tape library is the slowest but has the highest affordable capacity while the soft storage has the lowest affordable capacity but is the fastest. This suggests that most frequently requested objects must be stored in the soft storage while the least frequently used or requested files must be stored on the tape library.

Each continuous media file claims a portion of two important assets of the server: its storage and its bandwidth. Thus these files can be characterized by two parameters, namely; duration, which is a measure of how much storage a file requires and demand, which is the number of simultaneous streams requested from the same file. Consider a server with a total of F video files. Let Q_k denote the demand for the kth video file in number of streams and D_k the duration of the kth video file in seconds. The request arrival rate per second for the kth video file, λ_k, is given by:

$$\lambda_k = \frac{Q_k}{D_k} \tag{13}$$

The skew coefficient, θ, expresses the variation in demand between different multimedia files and varies between 0 and 1. If the probability, P_k, that an incoming request is for the kth video file is assumed to be a normalized geometric distribution then,

$$P_k = \frac{(1-\theta)\theta^k}{\theta(1-\theta^F)} \tag{14}$$

where k = 1,2,... F. If the total number of streams the server can support simultaneously is β_{TOT}, the demand Q_k for each video file is given by

$$Q_k = \beta_{TOT} P_k \tag{15}$$

In our design of a distributed multimedia system, the video server architecture will consist of multiple processors, a disk subsystem and a tape library. Each of the processors has its own cache and main memory or RAM, though they access the same disk subsystem and tape library through interfaces. The disk subsystem consists of multiple disks where each disk has one read/write arm and C_{DISK_i} GB of storage capacity. The disk subsystem has a total capacity given by:

$$C_{DISK} = B_{MULT} D_{DISK} \tag{16}$$

where D_{DISK} is the total duration of the video files stored on the subsystem in seconds. The arm of each disk has the capacity of reading a maximum of β_{ARM} video streams without jitter. This capacity is dependent on the compression technology used for video and audio, and the bandwidth capacity of the arm. If the bandwidth of the streams is B_{MULT} in bits/second and of the arm is B_{ARM} then

$$\beta_{ARM} = \frac{B_{ARM}}{B_{MULT}} \quad (17)$$

Disk stripping is a technique that increases the video file transfer rate for existing drive technologies. If a video file is placed onto a single disk without stripping, then the maximum number of simultaneous requests the server can satisfy is limited by the number of streams an arm can read without jitter. By stripping a file onto multiple disks, the server can satisfy more simultaneous requests for the same file. If the stripping factor is s_f, then the video file is distributed across s_f disks. This means that $s_f \beta_{ARM}$ simultaneous streams can be read out of a file.

The main memory or RAM can satisfy all requests as long as the total requested bandwidth does not exceed the system bus bandwidth i.e. $B_{MEM} = B_{SERVER}$. Thus the number of streams that can be delivered by the memory is given by

$$\beta_{MEM} = \frac{B_{MEM}}{B_{MULT}} \quad (18)$$

This memory is ideal for most frequently requested multimedia objects. The capacity of the memory is given by

$$C_{MEM} = B_{MULT} D_{MEM} \quad (19)$$

where D_{MEM} is the amount of video in seconds stored in memory.

The tape library consists of video storage shelves, tape loading mechanism and tape drives. The number of streams supported by the tape subsystem depends on several factors, such as the number of tape drives, the storage capacity of the cartridges, speed of the loading mechanism and the interface to the main system. The bandwidth capacity of the tape system is limited and hence the number of requests for the video files on tapes must be low so as to have acceptable delays. The tape system stores a backup copy of each multimedia file. If the tape library receives λ_T requests per second and the average length of a multimedia stream is D_{MULT} seconds then the number of simultaneous streams supported by the tape system is

$$\beta_{TAPE} = \lambda_T D_{MULT} \tag{20}$$

The requests are queued and served on a first come first served basis. The loading mechanism is the server and the service time is defined as the time taken to find the tape and load it into a tape drive.

The server migrates the less requested files from the tape drives to the disks and stores them using the stripping mechanism. Then the file is loaded into memory or RAM for distribution to the clients. If the file is one that is requested more often then the file is loaded into RAM from the disks and then transmitted to the client. Thus the storage hierarchy in terms of multimedia clips is $D_{MEM} \varepsilon D_{DISK} \varepsilon D_{TAPE}$.

Chapter 4

Mapping The Integrated Approach To A Case Study

4.1 NASA Classroom of the Future Program Testbed

An integrated computer-video local area network (C/V LAN) is designed and implemented as part of this research at the Center for Educational Technologies for the NASA Classroom of the Future Program. The LAN is installed using NASA support to facilitate the COTF Programs' mission of enhancing the learning and teaching process for mathematics, science and technology education using advanced computer and telecommunications technologies.

The Center for Educational Technologies has nine major functional areas:

- 21st Century Learning Center is a multimedia, multipurpose room with 30 multimedia-capable computers that run both Macintosh and PC based applications. These computer platforms have been selected because they are the most commonly found computers in K-12 schools. The room is also equipped with an instructor's workstation, three remote controlled cameras, two overhead high resolution video/data projectors and a voice-lift system.

- Cooperative Learning Center is where applied research is conducted on collaborative learning environments. It contains eight computers with multimedia authoring or developing capability with video and audio capture boards, laser disc players, CD-ROM drives and access to multimedia servers. This area is called a Cooperative Learning Center as the participants can be divided into groups of three and work at each station. Each member is assigned a task in the development of a multimedia presentation.
- Distance Learning Center is equipped with an instructor's workstation, three remote controlled cameras, a video/data projector, a visualizer, CD-ROM, laser disc, VCR and an audio (speaker and microphone) system for an audience of 35 people.
- Experimental Laboratory is a general purpose science lab. This area has three computers connected to probes for science experiments. Cameras have been provided to capture these experiments and broadcast to the other rooms.
- Discovery Center is a "hands-on minds-on" math/science experience room. The center includes highly interactive multimedia programs. A virtual reality simulation will provide three-dimensional depictions of objects and concepts that are difficult, if not impossible, to fully comprehend in two-dimension. The center also has a weather station where visitors can see "live" weather images, track

hurricanes across the oceans and land, look for weather fronts, study weather patterns, etc.

- NASA Regional Teacher Resource Center is the means of disseminating NASA educational materials to the public, to students and to educators. The center includes computer access to "on-line" resources and a multimedia software preview center in addition to reproduction facilities for print materials, software and video tapes.
- Challenger Learning Center consists of two separate facilities, a "Mission Control" and a "Spacecraft Simulator". This simulator is used for up to 40 students or adults. Following an orientation briefing, half the group proceeds into Mission Control and the other half through the airlock and into the Spacecraft Simulator. Both groups interact with each other using a computer network and a video system.
- Software Development consists of the software developers for the various multimedia packages developed for NASA. The developers are provided with high-end graphic and multimedia authoring workstations equipped with large memory capacities and high speed connections to the LAN.

- Administration comprises of the staff required for maintaining and operating the program and facility. All administrative staff have been provided with computers connected to the LAN.

4.2. Mapping Network Load Characterization to a Real-time Problem

In the previous chapter, we looked at a mathematical model for representing the traffic on any subnetwork or even the backbone. If we were to consider equation (11) (from section 3.4) and substitute

$$N^{TOT} = \sum_{i=1}^{S} N_i^{SUB} \tag{21}$$

we have

$$T^{BACK} = (1-\alpha)\,\beta\, N^{TOT}\, B_{MULT} \tag{22}$$

If we consider that only some of the users i.e δ are multimedia capable then

$$T^{BACK} = (1-\alpha)\,\beta\,\delta\, N^{TOT}\, B_{MULT} \tag{23}$$

Let us use this equation for a video conferencing scenario where the bandwidth for the multimedia streams is about 384 kb/s. Each user station may participate with 3 other users on the same network but may be different subnetworks i.e β = 4. Assuming that half the total number of users will have the capability to participate in the conference we have the values given in Table 4.1

Table 4.1 : Backbone Bandwidth Requirements for Video Conference Application

Case	(1-α)	N^{TOT}	T^{BACK} Mbps
1	0.2	50	7.68
2	0.4	200	61.44
3	0.6	600	276.48
4	0.8	1500	921.60

If we were to consider a video server delivering a broadcast of one video stream to every user on the network then the backbone capacity would change as shown in Table 4.2 as δ=1, β=1 and say B_{MULT}=1.33 Mbps:

Table 4.2 : Backbone Bandwidth Requirements for Client/Server Application

Case	(1-α)	N^{TOT}	T^{BACK} Mbps
1	0.2	50	13.3
2	0.4	200	106.4
3	0.6	600	478.8
4	0.8	1500	1596

Further, if we were to consider the video server distributing a live broadcast and say half the number of users were participating, in small groups of 4, in video conferences with each other regarding the broadcast then B_{1MULT}=1.33 Mbps, β_1=1 and B_{2MULT}=384 Kbps, β_2=4. The backbone capacity will increase considerably as shown in Table 4.3.

Table 4.3 : Backbone Bandwidth Requirements for Simultaneous Operation of Client/Server Multimedia and Video Conference Applications

Case	(1-α)	N^{TOT}	T^{BACK} Mbps
1	0.2	50	28.66
2	0.4	200	229.28
3	0.6	600	1031.76
4	0.8	1500	3439.20

The above three tables show that with small number of workstations on the network we can use ethernet, fast ethernet or FDDI network architectures but as the numbers increase ATM and switching technology need to be considered. The high-speed and high-capacity backplanes of the switching hubs can sustain the bandwidth demands of the multimedia based activity on the network.

4.3 Multimedia Server Implementation

The video server installed in the C/V LAN is a standard of the shelf product called "StarWorks". The system is based on the media transport protocol (MTP) which is designed for continuous media or streams of video and audio [49]. The protocol accomplishes bandwidth reservation on the network interface of the server to ensure quality of service. Long messages or streams are optimized during delivery. The protocol [7] also provides control of the rate of transfer and also supports multicast connections. The server as seen in Figure 4.1, performs stream management in order to service users who simultaneously recording or playing back video files. This management also allows access to multiple files. These files may be delivered at different rates of transfer depending on the original recording or the negotiations with the user. The stream management system also permits addressing of the multimedia stream either based on time or frame.

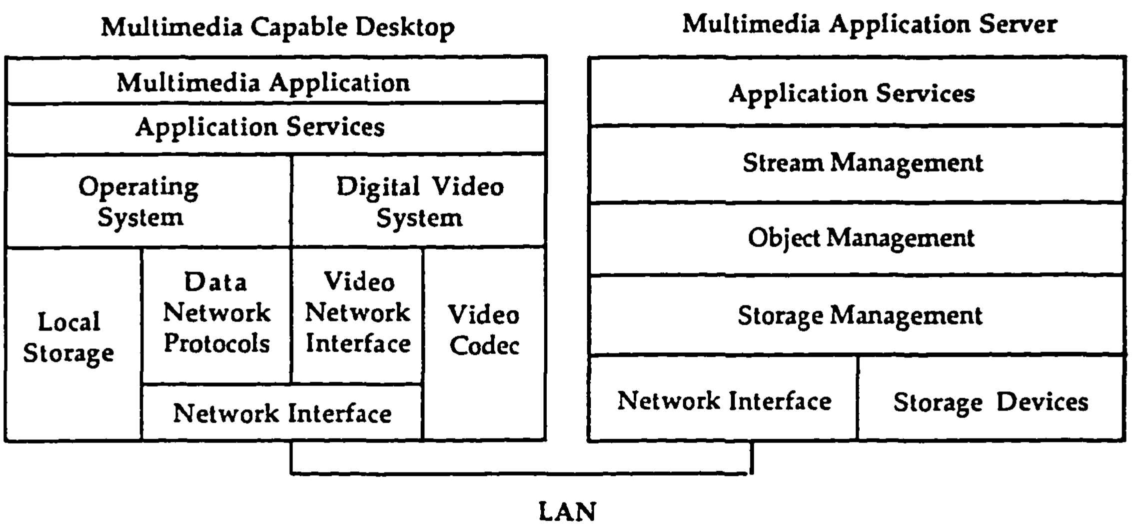


Figure 4.1 : StarWorks Client/Server Architecture

During implementation it was necessary to decide on the type of platform that was to be implemented for the server. Based on technical specifications, expandability and flexibility a Sparc Server 1000 with at least 6 processors was selected. The multiple processors would be utilized to service a large number of users (approximately 300) simultaneously. Each processor would be able to service about 50 streams concurrently. One of the critical factors in this implementation was to determine the size of the storage components based on our discussion in section 3.4.

- If it is assumed that 10 seconds of MPEG compressed video is pre-staged in the RAM to service requests then the RAM modules for the memory must have a total capacity which is given by

$$C_{MEM} = B_{MULT}\, D_{MEM} = 1.5 \text{ Mbps} \times 50 \text{ streams} \times 10 \text{ sec/stream} \approx 94 \text{ MB}$$

Thus the capacity of the memory modules on each processor must be between 96 MB and 128 MB. This would be sufficient to service 50 streams of MPEG compressed data to clients without any delay. The server bus, which is an Sbus, has a capacity of 800 Mbps and hence the actual capability of these RAM modules is

$$\beta_{MEM} = \frac{B_{SERVER}}{B_{MULT}} = \frac{800}{1.5} \approx 533 \text{ streams}$$

But in order to service these number of streams the amount of memory will be very large. Also a server bus will hold two processors and hence there will be at least 50 other streams being transmitted through the bus.

- The disk capacity is calculated for the storage requirement of about 25 hours of MPEG compressed multimedia objects. This gives a total of

$$C_{DISK} = B_{MULT} D_{DISK} = 1.5 \text{ Mbps} \times 25 \text{ hours} = 16.9 \text{ GB}$$

If we take into account the space required for the server software and other maintenance programs we would require a total capacity of 18.9 GB of disk capacity. The disk subsystem would consist of multiple disks which were placed in a RAID 4 configuration. The stream data is striped across all disks so that maximum number of simultaneous accesses are possible. The stream data is grouped, sorted and read in a strict periodic fashion. The number of streams supported by these disk modules is given by

$$\beta_{DISK} = \frac{B_{ARM}}{B_{MULT}} = \frac{280}{1.5} \approx 187 \text{ streams}$$

The disks are attached to the server system by Fast SCSI-II interfaces which have a bandwidth of about 160 Mbps. Hence the total number of streams that can be supported by the disk drive is 100 per interface. If we use a total of 3 or 4 interfaces the server could support about 300-400 streams of MPEG compressed video.

• The tape library system will need capacity to store the 25 hours of video staged on the disk drives and an additional 40 hours of video that may be requested occasionally. Thus the capacity of the tape library must be

$$C_{TAPE} = B_{MULT} D_{TAPE} = 1.5 \text{ Mbps x } 65 \text{ hours} = 43.9 \text{ GB}$$

Due to manufacturing and cost restrictions if we were to use 5 GB tapes then the total capacity of the tape system will be about 45 GB. The number of streams supported by these tape system is given by

$$\beta_{DISK} = \frac{B_{ARM}}{B_{MULT}} = \frac{280}{1.5} \approx 187 \text{ streams}$$

4.4. Computer/Video LAN Implementation

The layout of the computer-video LAN implemented at the NASA Classroom of the Future Program has been designed taking into consideration all the issues raised in the previous sections. The network is one of the few networks installed in a production environment that integrates the computer and the video network. A diagram showing the different components is shown in Figure 4.2.

The computer network will serve approximately 50 multimedia workstations capable of running both Macintosh based and PC based applications, 75 other workstations used for administration, software development and multimedia authoring. The network is based on switched ethernet and ATM technology. The decision to use these technologies is based on the capacity calculations discussed in the previous section. The calculations were based on the type of activities taking place on the network. The resources on the network include an internet server, a video server with its large storage and a Novell application server. The network was designed on the premise that the servers would have high bandwidth connections to the network and the workstations were serviced by dedicated bandwidth connections.

Compression techniques, discussed in the previous chapter, such as MPEG, QuickTime are being employed to distribute video and audio data. The network also requires synchronization features, guarantees for quality of service and congestion control to maintain the bandwidth for the distribution of digital continuous media.

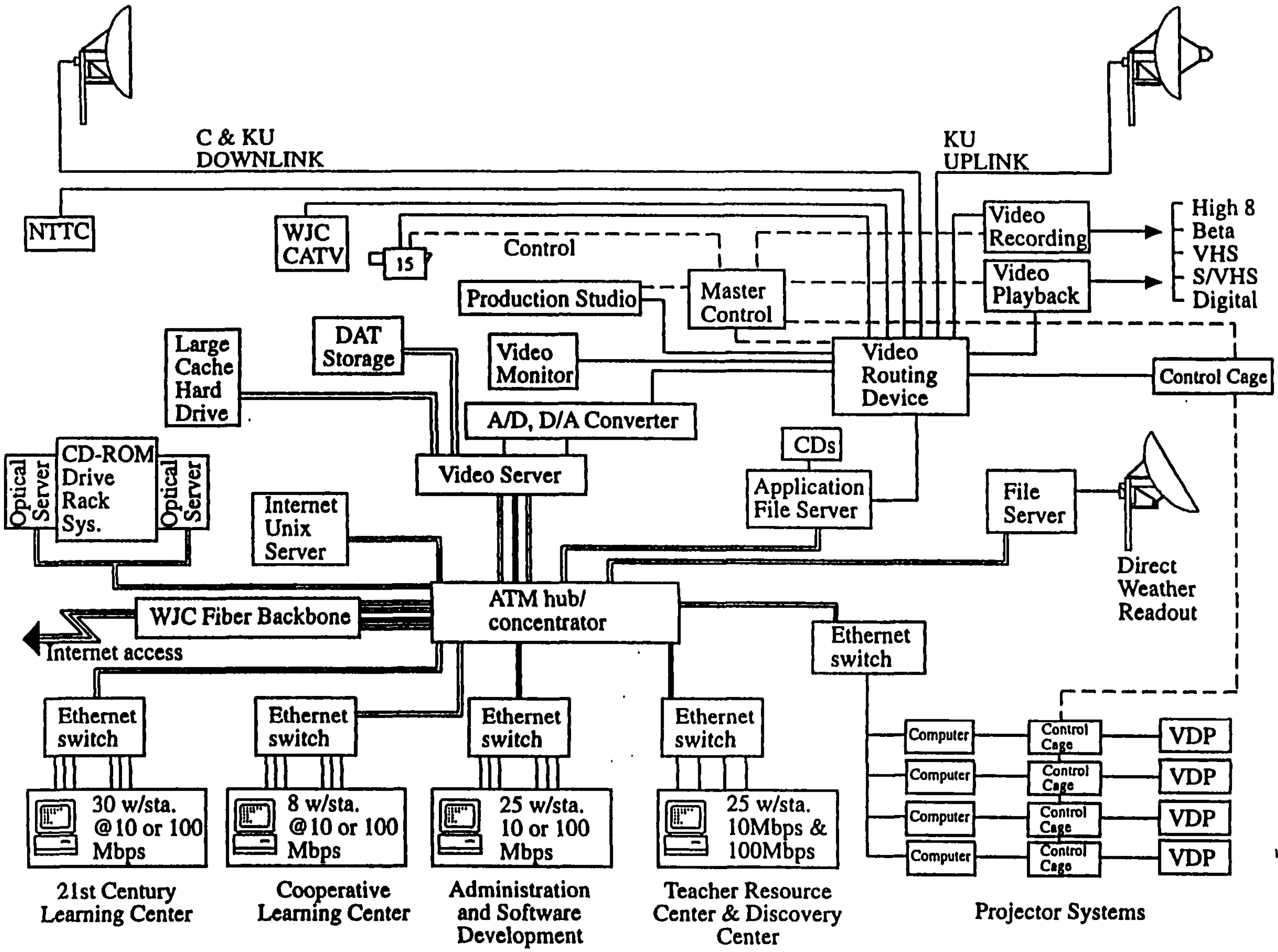


Figure 4.2 : Computer/Video Local Area Network

The video network consists of a number of video recording, playback and display devices. It also includes the audio components required in such a system. The facility is equipped with a television studio, a production control area, an offline editing suite, a Ku-band satellite uplink dish, two C/Ku-band downlink dishes and two weather station receiving antennae. Each of the rooms described in the earlier section have equipment and cameras which are connected to the video network for transmission of video signals and for control of the cameras. The equipment are controlled by a card cage and computers which are connected to the computer network. The routing of the video and stereo audio signals is achieved with video routers and a software with a GUI interface running on the control computers.

The video and audio signals are routed to the digitizing workstations for broadcast by the video server or storage in the video server for later use. The digitizing stations compress the video and audio to either QuickTime, Video for Windows or MPEG format for playback on the multimedia workstations in the various rooms and for use by the software developers. The reverse action may also be achieved by performing an analog conversion of the digitized animation from the computer network to the video network for use in video production or for broadcast using the teleport facility. Thus, there is total integration of the two networks.

4.5 Benchmarks for the Operation of the Integrated Network

As part of this research, it is important to gather information of the operation of the network described in the previous section. This information is basically a benchmark for the state of the network while users are operating in the distributed multimedia environment. The scenarios considered in the establishment of benchmarks comprised of:

- a workstation which digitizes the video and audio in real-time and broadcasts it across the network for a video conference situation;
- clients access video files that are already digitized and stored on the multimedia server which has two 10 Mbps ethernet connections; and
- a combination of the above two i.e. clients that are watching the video conference broadcast and viewing video files from the server.

It is important to consider that the network topologies used in these scenarios were either a shared media hub, or a switching hub connecting two shared media hubs or two switching hubs with an ATM backbone. Thus the clients were either sharing 10 Mbps between each other or had dedicated 10 Mbps links.

<u>Scenario 1</u>: Six workstations were connected to a shared ethernet hub as shown in Figure 4.3. One of the stations was broadcasting or could be considered as the main station for the video conference. The remaining five stations were watching this video conference broadcast.

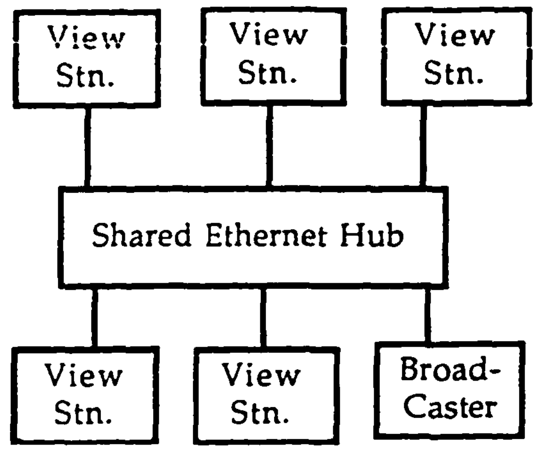


Figure 4.3 : Topology 1

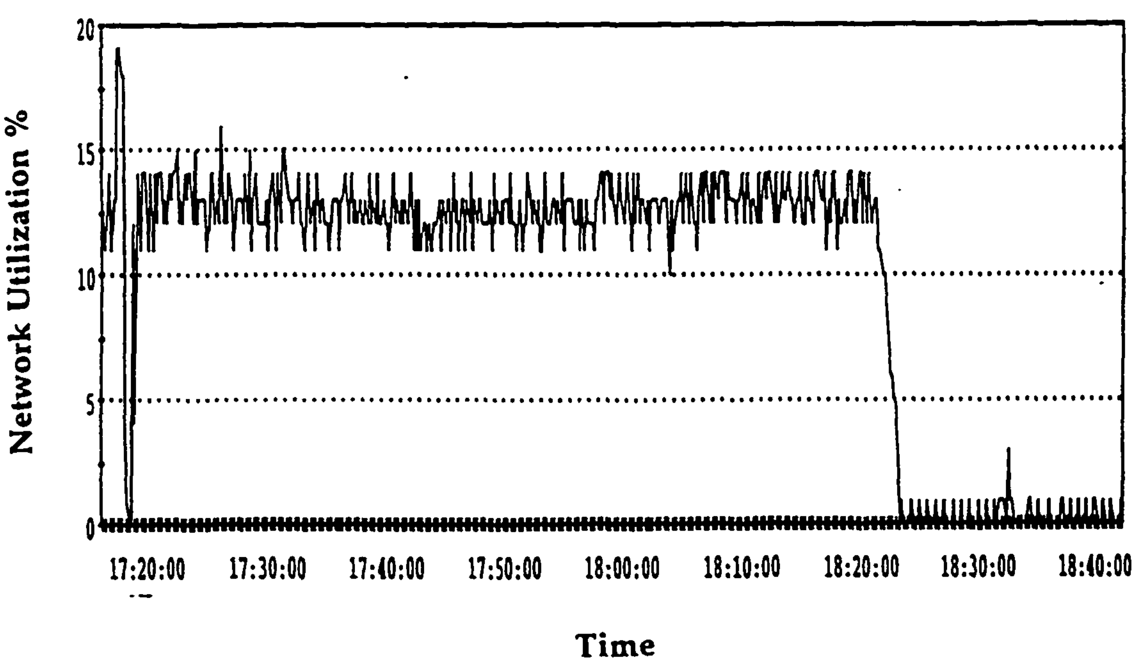


Figure 4.4 : Network Utilization for Topology 1

As can be seen from the graph shown in Figure 4.4, the network utilization increases from 0 to about 13% while the clients open their channels to view the broadcast. The frame rate of the broadcast was 15 frames per second.

Scenario 2: Five stations were setup as client stations to the multimedia server on a shared 10 Mbps ethernet segment or hub. As mentioned before, the video server stores and forwards the video and audio files to the clients, unlike the conference broadcaster which does not store the video. The server had two connections to the ethernet segment. The setup is shown in Figure 4.5.

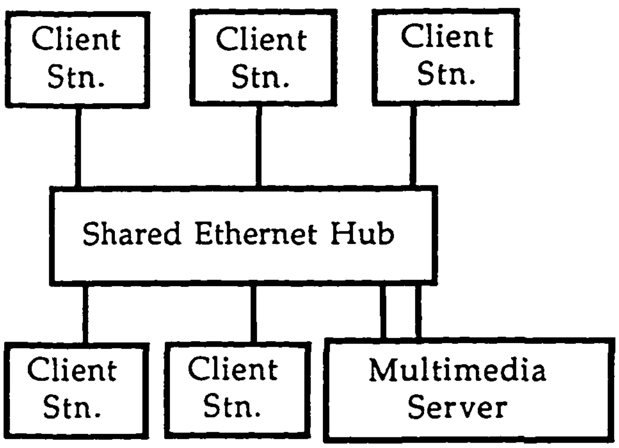


Figure 4.5 : Topology 2

Initially, four of the five clients were interacting with the same multimedia clip which included text, graphics, animation, video and stereo audio. The clip had a total size of 46.1 MB and distributed by the server at 2.23 Mbps. The movie resolution was 316x236 and of about 3 minutes and 2 seconds in duration. The movie was distributed at 30 frames per second. When the fifth client was brought on to interact with the server, it was observed (Figure 4.6) that the system underwent rapid degradation leading to an eventual crash of the system. In the graph the peaks and troughs indicate the interaction of the clients with the video streams using commands such as fast forward or rewind or pause.

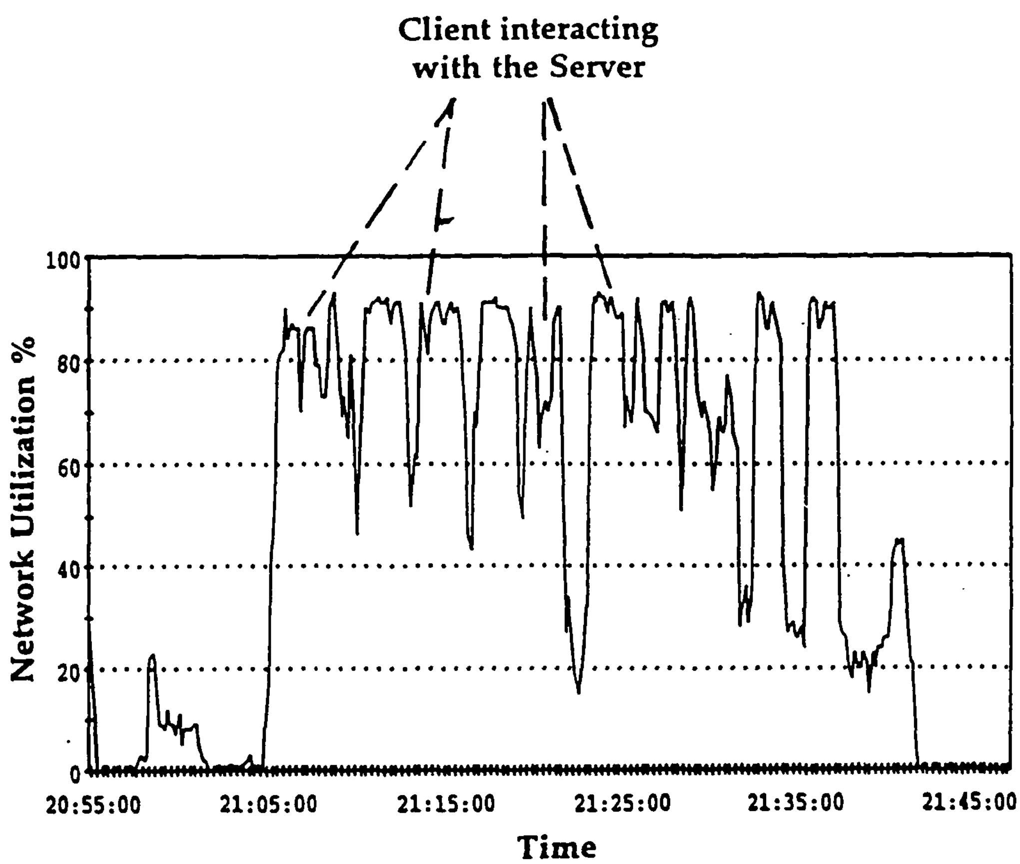


Figure 4.6 : Network Utilization for Topology 2

Scenario 3: Eight stations were connected to the multimedia server via the shared ethernet hub. In this scenario the clients watched a video clip which had been MPEG compressed. The clip had a resolution of 240x180 and was of 2 minutes duration. Table 4.4 shows the bandwidth being put out by the server, CPU utilization of the server and the bandwidth utilization of the network. The table also shows the frames per second being played on each workstation as additional users watched the same clip.

Table 4.4 : Observations from Topology 3

Number of Stations	BW from Server (Mbps)	Server CPU Utilization %	Frames per second	Network Utilization %
1	1.33	4	30	13
2	2.66	10	30	30
3	3.99	15	30	43
4	5.22	18	24	60
5	6.55	24	24	65
6	7.88	29	15	78
7	9.11	34	10	90
8	10.44	40	8 - 0	98 - 0

When all eight stations were trying to watch the video file it was observed that initially the frame rate dropped to 8 frames per second for a short period and then the network hung till some of the clients were turned off and the bandwidth in the segment was recovered. The graphical output of the monitoring station is shown in Figure 4.7. The graph shows interactions by the clients with the clip by the dips and peaks, as the interactions cause the server to stop sending data and respond to the request.

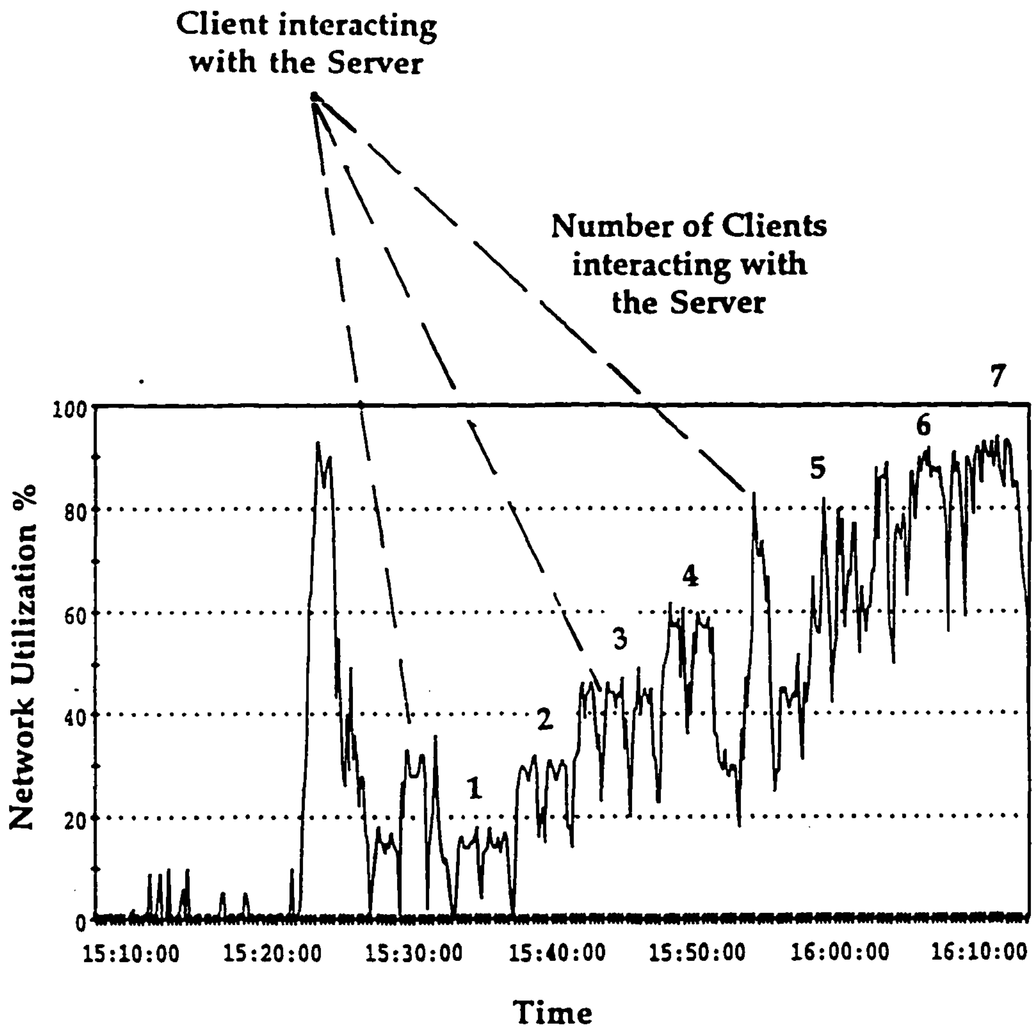


Figure 4.7 : Network Utilization for Topology 3

Scenario 4: This scenario was a combination of scenarios 1 and 3. In this scenario, 3 stations were setup to tap into the video conference broadcast and four stations are setup to view the same video file from the server. All the seven stations, the station that broadcasts and the multimedia server were all connected to the same segment. Hence the 10 Mbps was shared by all stations.

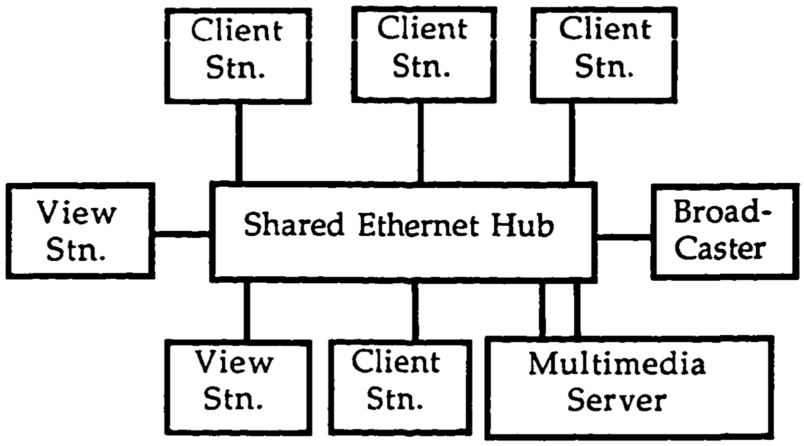


Figure 4.8 : Topology 4

The clients viewed a video clip which was delivered to each one at 1.33 Mbps. The resolution of the video was 240x180 and the clip was 2 minutes

Table 4.5 : Observations from Topology 4

VC BROADCAST		CLIENT / SERVER			
Number of Stations	Frames per second	Number of Stations	Frames per second	Server CPU Utilization %	Bandwidth Utilization %
1	30	0	0	4	4
2	24	0	0	4	7
3	18	0	0	4	9
3	15	1	30	11	25
3	12	2	30	18	39
3	10	3	24	24	52
3	6	4	20	28	70

duration. Graph in Figure 4.9 shows the network utilization as the different stations were activated on the network. Table 4.5 shows how the receiving frame rate at the broadcast viewing stations reduced as the number of stations active on the segment increased during the experiment which is attributed to the amount of bandwidth available to the systems.

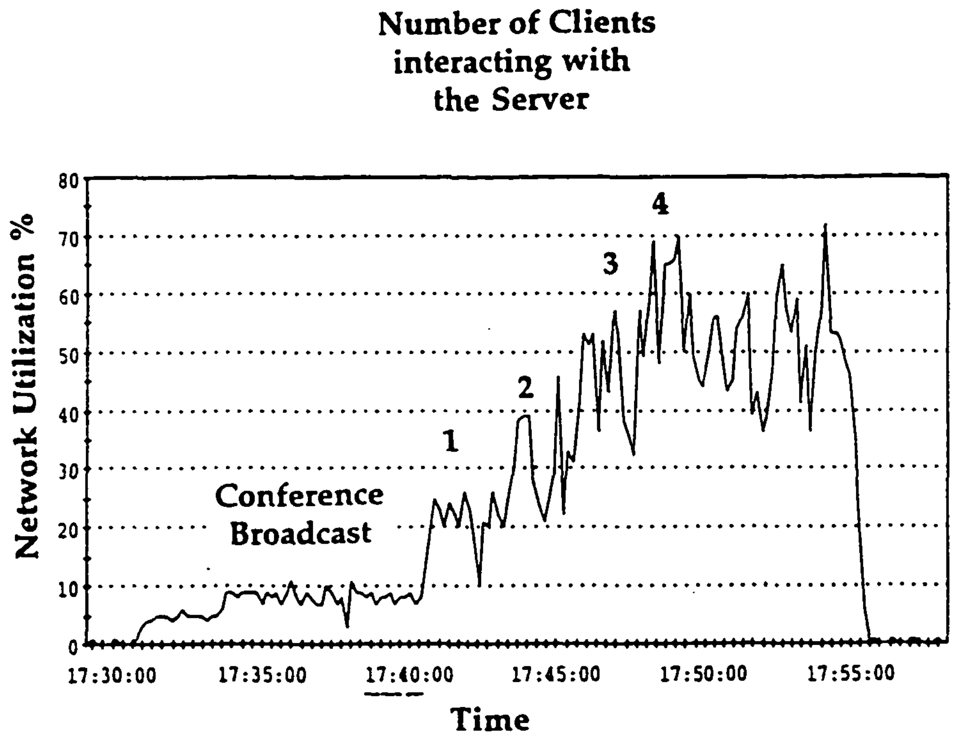


Figure 4.9 : Network Utilization for Topology 4

Scenario 5: In this scenario, we considered two shared ethernet segments connected by a switching hub as shown in Figure 4.10. This guaranteed 10 Mbps between the hubs. The stations were connected to the conference broadcaster one at a time alternately between the two hubs. It must be noted that the video conference broadcaster was connected only to one hub. The broadcast was at 240x180 resolution and was distributed at 0.5 Mbps.

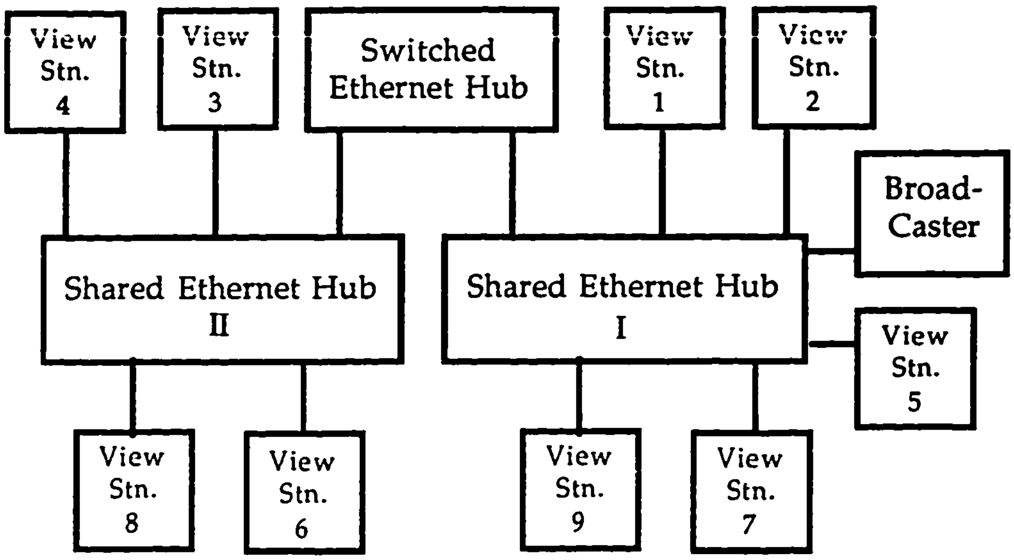
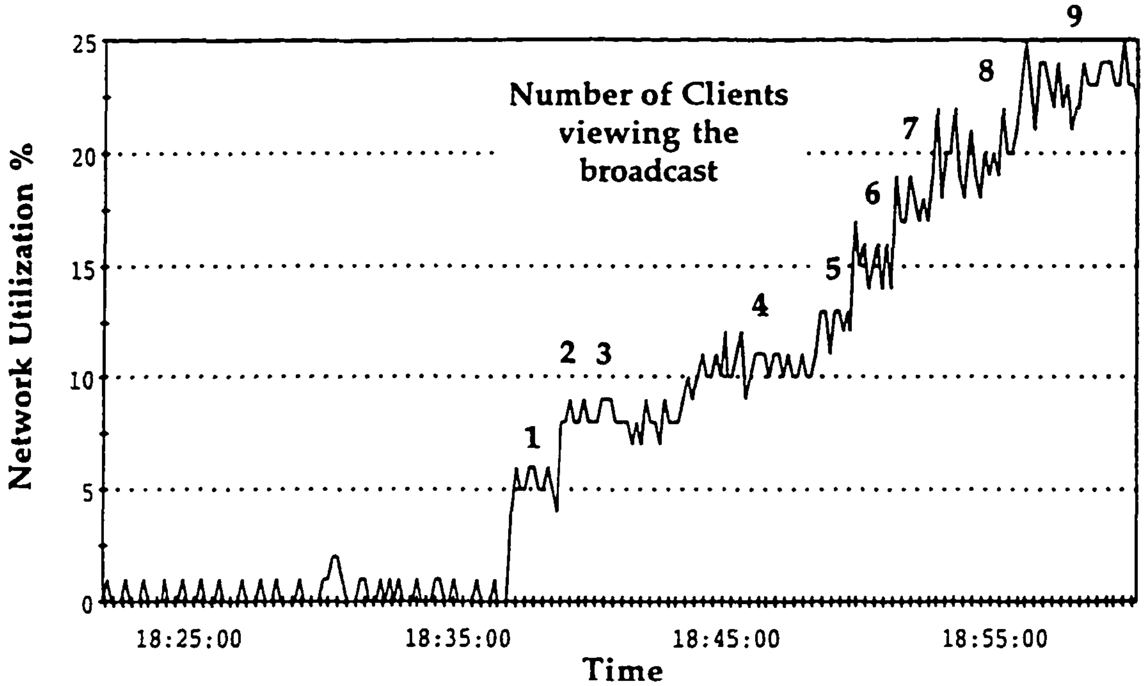


Figure 4.10 : Topology 5

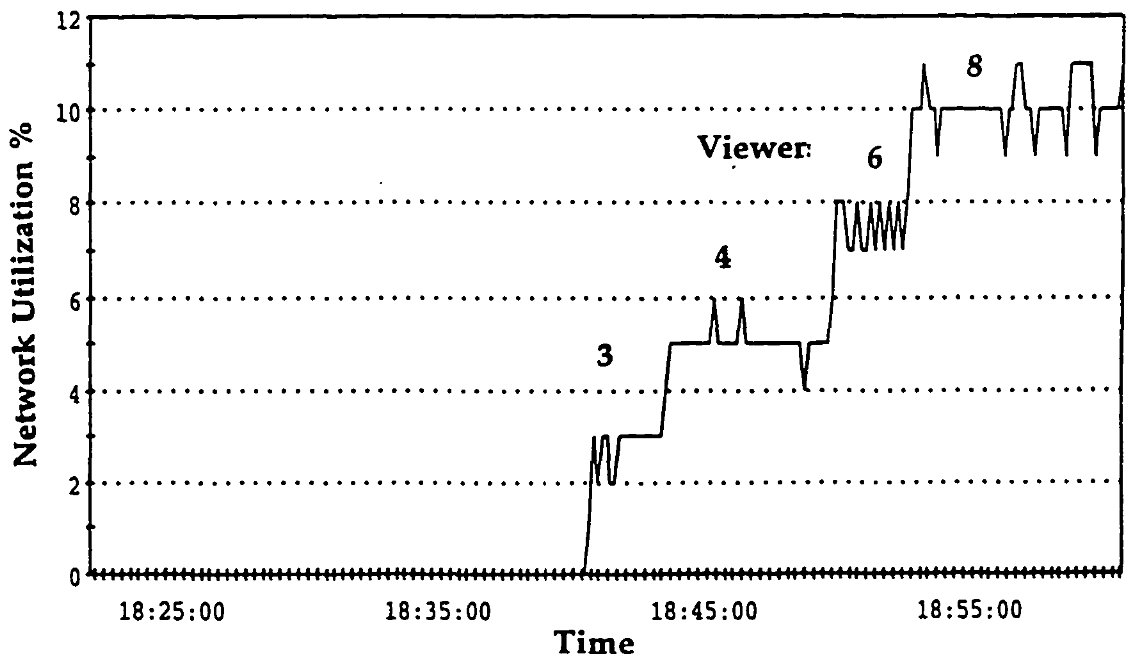
As we see in the graphs (Figure 4.11) network utilization in hub I kept increasing as each viewing station connected to the conference broadcaster from 0% to almost 25%. The utilization in hub II increased from 0% to 11% as the four stations connected to the broadcaster. It is equally important to note that as the number of stations connecting increased, the frame rate of the broadcast on the viewing stations decreased (Table 4.6).

Table 4.6 : Observations from Topology 5

HUB I		HUB II	
Number of Stations	**Frames per second**	**Number of Stations**	**Frames per second**
1	15	3	12
2	15	4	10
5	10	6	8
7	8	8	6
9	6		



Hub I



Hub II

Figure 4.11 : Network Utilization for Topology 5

Scenario 6: This scenario is similar to the above scenario except the application being viewed is of client/server type. The clients interacted with the same video file from the server which was being delivered at 1.33 Mbps with a resolution of 240x180 and the clip was of 2 minutes duration.

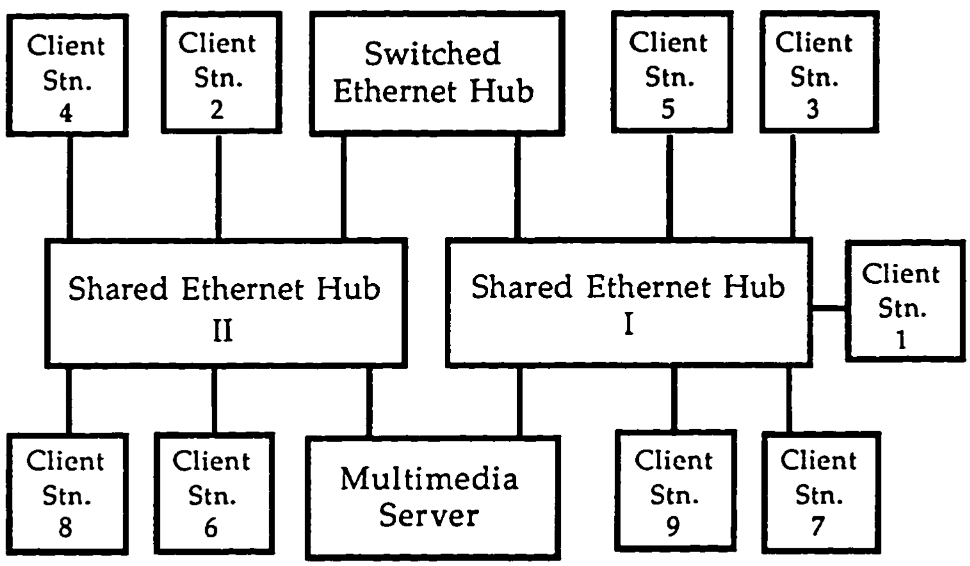
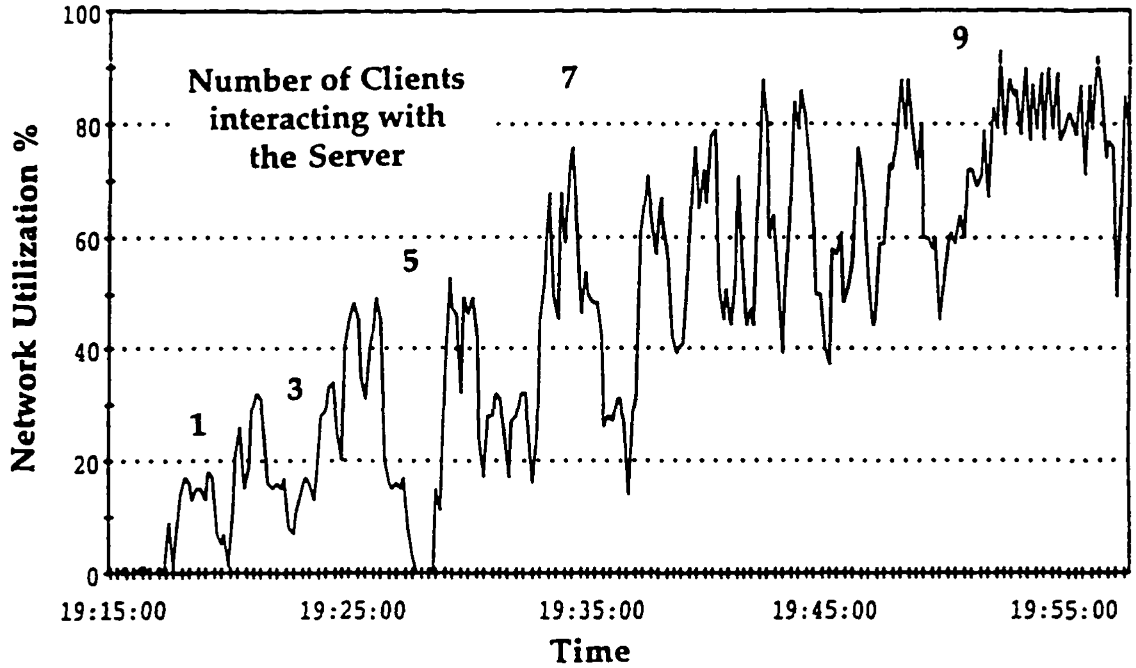


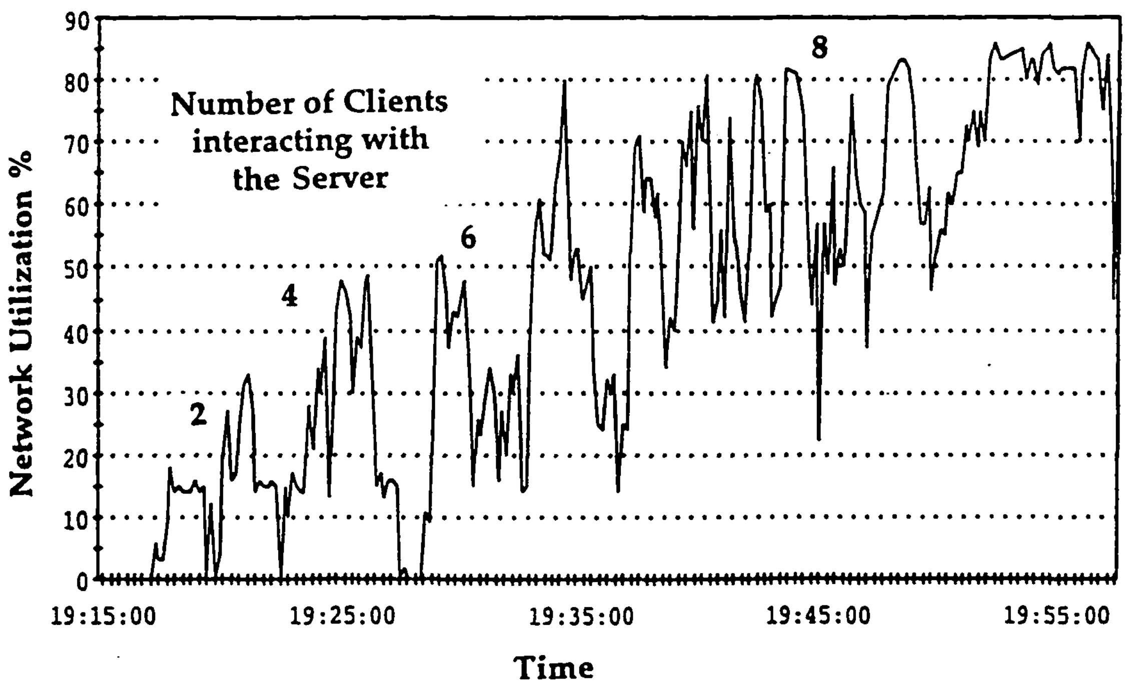
Figure 4.12 : Topology 6

Table 4.7 : Observations from Topology 6

HUB I			HUB II		
Number of Stations	**Frames per second**	**Server CPU Utilization %**	**Number of Stations**	**Frames per second**	**Server CPU Utilization %**
1	30	11	2	30	18
3	30	24	4	30	28
5	30	32	6	26	36
7	22	40	8	18	42
9	15	45			



(a) Hub I



(b) Hub II

Figure 4.13 : Network Utilization for Topology 6

As can be seen from Table 4.7 and the graph in Figure 4.13, the number of clients interacting with the server and requesting to view the stored video file increased CPU utilization, increased the network utilization, but reduced the frame rate. This reduction in frame rate is due to the limited bandwidth available on each of the segments and not on the capability of the server as CPU utilization was below 50%.

Scenario 7: In this scenario we had clients and viewing stations attached to both hubs, the video conference broadcaster is attached to only hub I and the multimedia server is attached to both hubs as shown in Figure 4.14.

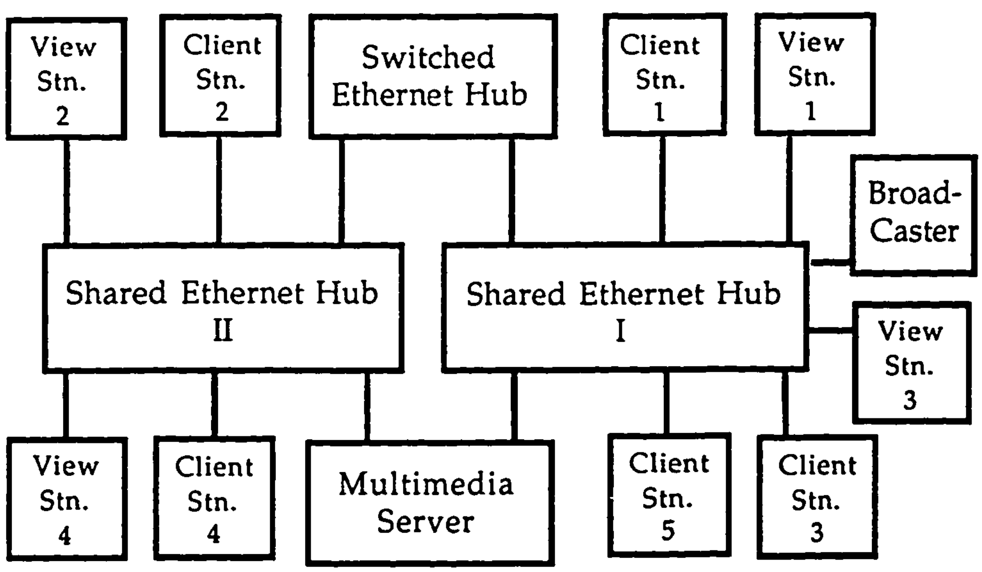
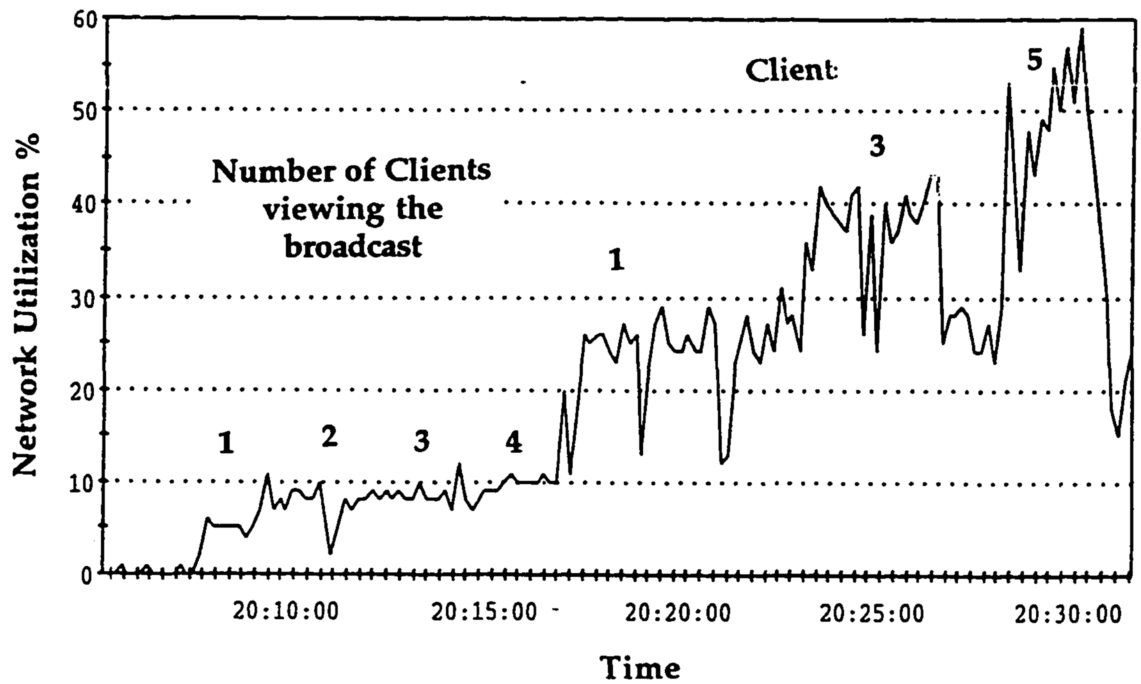
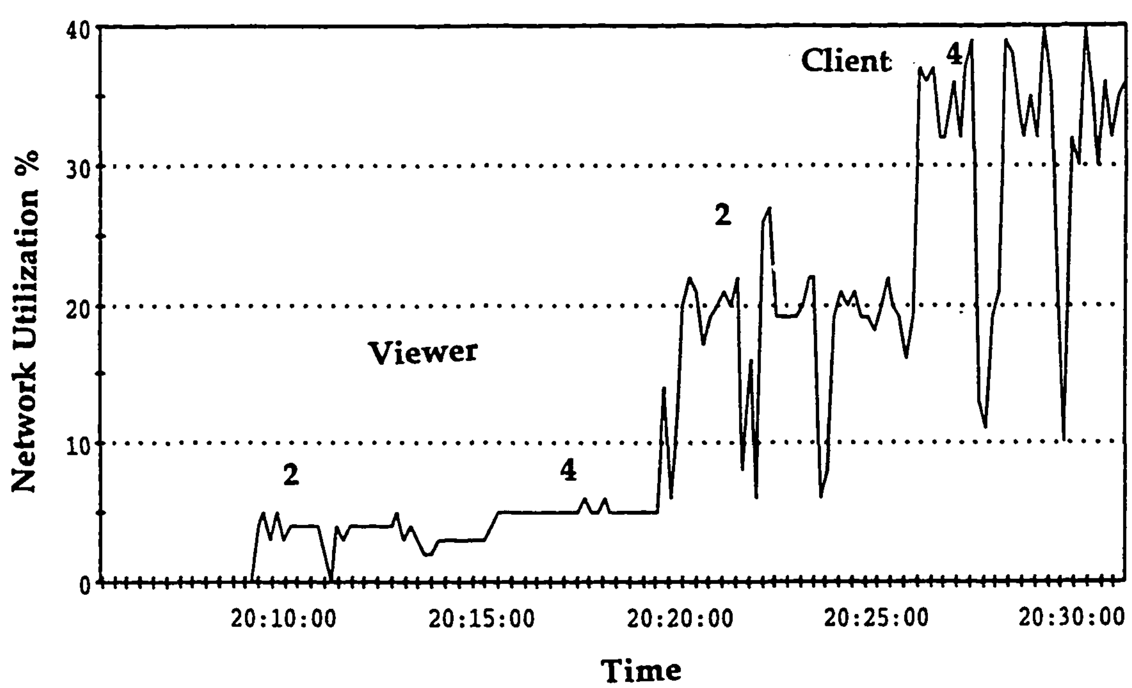


Figure 4.14 : Topology 7

Table 4.8 shows that the frame rate for the clients did not reduce but the rate for the broadcast viewing stations reduced as more clients interacted with the video file on the server. The network utilization graph in Figure 4.15 indicates that the utilization did not increase with the number of



(a) Hub I



(b) Hub II

Figure 4.15 : Network Utilization for Topology 7

viewing stations but increased as more clients interacted with the server. It may also be noticed that the utilization on hub I increased significantly as the number of viewing stations connecting to the broadcaster increased.

Table 4.8 : Observations from Topology 7

VC BROADCAST		CLIENT / SERVER		
Number of Stations	**Frames per second**	**Number of Stations**	**Frames per second**	**Server CPU Utilization %**
1	15	0	0	4
2	15	0	0	4
3	12	0	0	4
4	10	0	0	4
4	10	1	30	11
4	8	2	30	18
4	6	3	30	24
4	5	4	30	28
4	4	5	30	32

This increase is due to the fact that all requests and video data for hub II viewing stations travels through hub I. But if clients on hub II were to interact within themselves, then they would have bandwidth available to exchange data.

Scenario 8: This configuration is similar to scenario 5 except that the shared ethernet hubs are replaced by switched ethernet hubs as shown in Figure 4.16. A number of clients are connected using switched ethernet or dedicated 10 Mbps connections. The hubs are connected via an ATM backbone at 155 Mbps. Each client and the broadcaster are on their own segment with a capacity of 10 Mbps respectively.

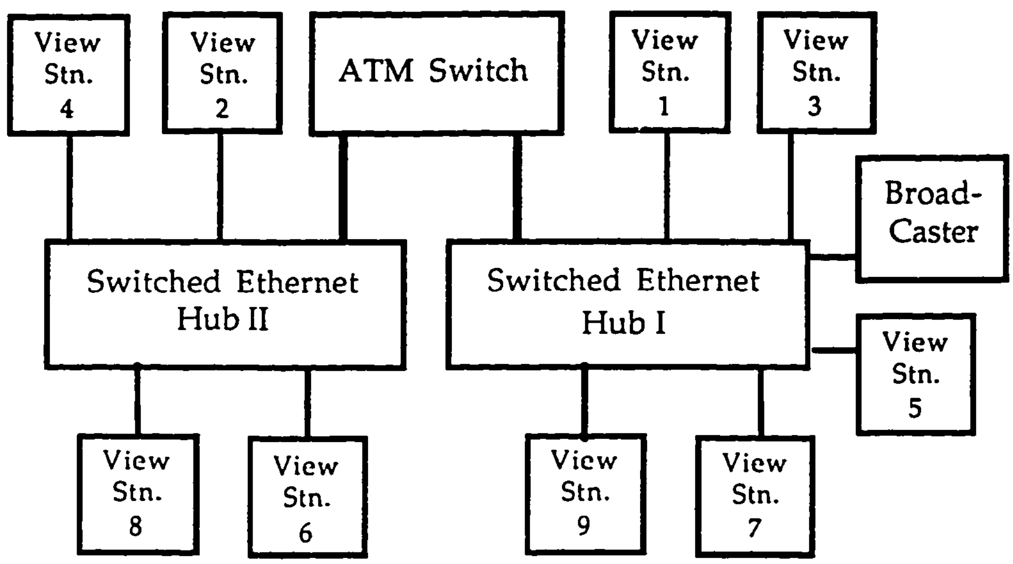


Figure 4.16 : Topology 8

Table 4.9 : Observations from Topology 8

HUB I		HUB II	
Number of Stations	**Frames per second**	**Number of Stations**	**Frames per second**
1	15	2	15
3	12	4	10
5	10	6	8
7	8	8	6
9	6		

It must be noted the observations in Table 4.9 show that the number of viewing stations is the same as in topology 5. The reason for this is that all the traffic to the viewing stations traverses the link between the switched ethernet hub and the broadcaster which has a limit of 10 Mbps.

Scenario 9: This scenario is similar to the above scenario except the application being viewed is of client/server type (Figure 4.17). The clients viewed the same video file from the server which was delivered at 1.33 Mbps and 240x180 resolution.

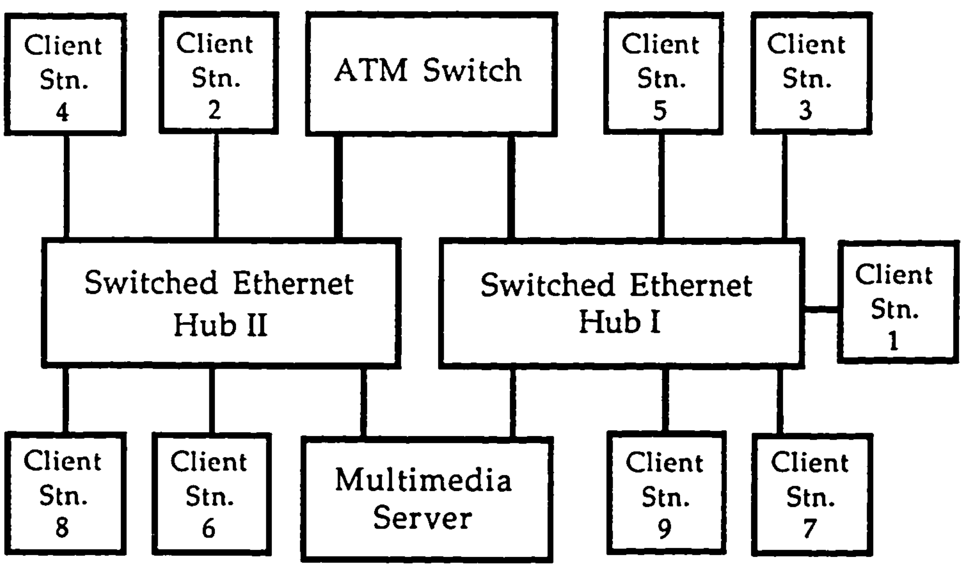


Figure 4.17 : Topology 9

As can be seen from Table 4.10, the number of clients interacting with the server and requesting to view the same stored video file increased CPU utilization, increased the network utilization but reduced the frame rate. This table is the same as Table 4.7 for topology 6.

Table 4.10 : Observations from Topology 9

HUB I			HUB II		
Number of Stations	**Frames per second**	**Server CPU Utilization %**	**Number of Stations**	**Frames per second**	**Server CPU Utilization %**
1	30	11	2	30	18
3	30	24	4	30	28
5	30	32	6	26	36
7	22	40	8	18	42
9	15	45			

Scenario 10: This scenario is similar to scenario 9 except that the multimedia server is connected via an ATM connection as shown in Figure 4.18.

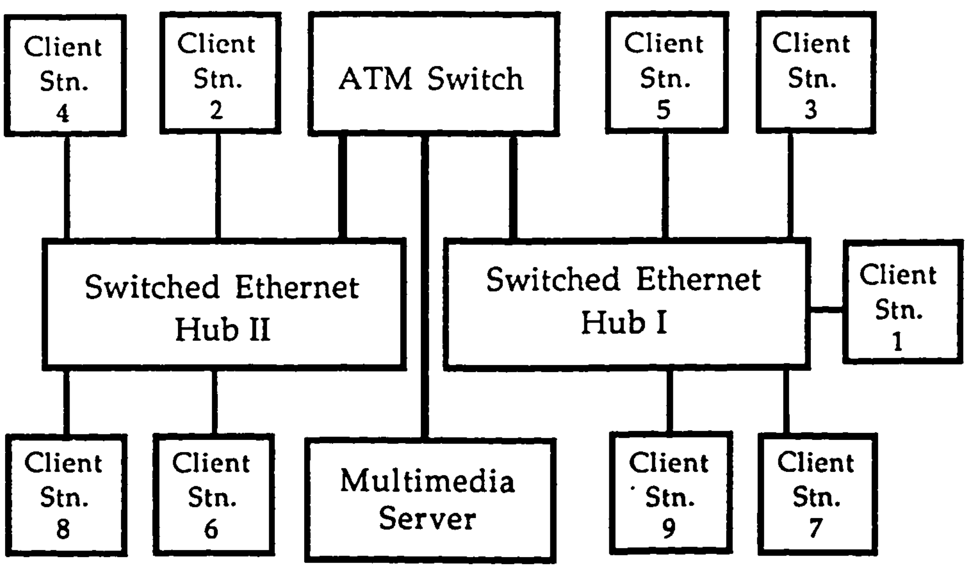


Figure 4.18 : Topology 10

As can be seen from Table 4.11, the number of clients interacting with the server and requesting to view the same stored video file increased CPU utilization but did not affect the frame rate. This is due to the fact that network utilization of each of the client links was either at 0% or was equal to the delivery rate of the video clip. Also, there is dedicated virtual circuit between the client and the server with bandwidth equal to the rate of the clip which resulted in viewing at 30 fps (Table 4.11).

TABLE 4.11 : Observations from Topology 10

HUB I			HUB II		
Number of Stations	Frames per second	Server CPU Utilization %	Number of Stations	Frames per second	Server CPU Utilization %
1	30	11	2	30	18
3	30	24	4	30	28
5	30	32	6	30	36
7	30	40	8	30	42
9	30	45	10	30	50
11	30	54	12	30	59
13	30	63	14	30	68
15	30	72	16	30	79
17	30	84	18	30	89
19	30	93	20	30	97

4.6 Discussion

Summarizing, the following observations were made during this benchmarking process.

1. As we increase the amount of bandwidth, by reducing the number of stations per segment, the total number of clients viewing a broadcast or interacting with a video clip from the video server increased.

2. It is further observed that as the bandwidth available on the network and the number of users increase, the bottleneck for multimedia traffic migrates from the network to the desktops, server or the broadcaster. This is due to the fact that additional processing power is required to keep up with the data or requests received from the network.

3. The operating system and legacy LAN protocols used in the network implement the best effort policy. As the number of clients or multimedia capable stations increase on the network, the number of simultaneous requests at the multimedia server increase or the number of clients watching a broadcast or participating in a video conference increase. This results in network bandwidth and system resources being divided to meet the demands of the clients, thereby causing in degradation in service level. In order to overcome this problem a quality of service component must be added to the server and station operating system to ensure that delivery of data simultaneously to some clients and not all clients at the requested or desired levels is maintained at all times.

4. Demand of bandwidth was determined by size of the video stream. The size of video stream is determined by the frames per second at which the video is recorded, the resolution and the number of colors used for depicting depth of picture. Increase in any one of these parameters for any one client results in need for bandwidth increasing quite rapidly.

5. The ideal scenario was 10 where the server had an ATM connection and the individual stations were connected via dedicated 10 Mbps ethernet connections. This is due to the fact that each client talking to the server had a dedicated connection throughout due to the establishment of a virtual circuit from the switching hub to the server. The bandwidth of this circuit was equal to the delivery rate of the stream. It must be noted that the applications and delivery of the video stream is still based on best effort. There needs to be mechanisms designed for guaranteed quality of service for the network and the server operating system.

6. If we were to compare the network utilization shown in scenario 6 and the data obtained from our computational model listed in Table 4.2 that they are almost identical for $1-\alpha = 1$ and $N^{TOT}=6$. Using equation (22) from section 4.2 we find that $T^{BACK}=7.98$ Mbps which leads to a network utilization of 79.8% but the actual utiization in hub I is about 95% or 9.5 Mbps. The additional 1.52 Mbps or 15.2% bandwidth is considered as the overhead of the ethernet transport protocol. Thus, the equation needs to be modified as:

$$T^{BACK} = (1-\alpha)\,\beta\, N^{TOT}\, B_{MULT} + \phi \qquad (24)$$

where ϕ represents the overhead required for the transport protocol as a fraction of the total bandwidth capacity required in the backbone.

Chapter 5

Establishment Of Guaranteed Quality Of Service Channels

5.1 Preliminaries

The implementation of the network architecture for a distributed multimedia environment described in the previous chapter was found, during the benchmark operations, unable to effectively use the ATM features such as transporting constant bit rate video and audio data. This drawback is mainly due to the fact that the network is a hybrid topology consisting of ATM and ethernet, and uses best effort policy. These types of networks will be implemented in the field for a long time due to the large cost factor that is encountered for a pure ATM network. Thus, to maximize this combination of network technologies we need to modify the operating system and the ethernet technology to handle multimedia traffic.

The multimedia operating system (MOS) must be able to request quality of service in the distributed environment. The QoS mechanism can be easily implemented in the ATM world but in the legacy LAN world it needs considerable thought. In this chapter we describe how these mechanisms can be implemented in the ATM and ethernet networks, respectively. Then we will consider how the mechanisms can be implemented in a heterogenous network consisting of ethernet and ATM technologies such as the C/V LAN implemented in this project.

5.2 Quality of Service (QoS) in a Distributed Multimedia Environment

The OSI reference model has a number of QoS parameters describing the speed and reliability of transmission, such as throughput, transit delay, error rate and connection establishment failure probability [12, 22]. These parameters apply mainly to the lower protocol layers and are not meant to be accessible by the application or the user. Further, these parameters are suitable only for the traditional data applications which are not time-sensitive. The parameters are not applicable in a distributed multimedia environment.

Quality of service represents the set of those quantitative and qualitative characteristics of a distributed multimedia system necessary to achieve the required functionality of an application [2, 3, 9]. Functionality includes both the presentation of multimedia data to the user and general user satisfaction. All components of a distributed system have their own QoS parameters. Some of these parameters possess dependencies which are expressed by mappings between the system's architectural layers. An application process originates the QoS requirements and conveys them in the form of QoS parameters to the other system components. Generally, a negotiation process among the components of the system must determine if the system as a whole can satisfy the requested QoS level. Different applications on the same distributed system can have different subsets of relevant QoS parameters with different required values, but some parameters might not be mutually independent.

Processing QoS in a distributed multimedia system involves several related activities:

- Assessing the QoS requirements in terms of users' subjective wishes or satisfaction with the quality of the application in terms of performance, synchronization, clarity, etc.
- Mapping the assessment results onto QoS parameters for various system components or layers. For example, the user chooses video in terms of its resolution and frame rate, which map onto throughput requirements.
- Negotiating between system components or layers (embedded in protocols) to ensure that all system components can meet the required parameters consistently. If the negotiation ends with an agreement on the required values, the application can be launched. Types of agreements include guaranteed, best-effort or stochastic.

In practice, the required QoS values corresponds to an instantaneous operating point within the parameter space. This operation point may change over time within a region in this space causing a renegotiation of the QoS contract. Sometimes the negotiated parameters cannot be maintained due to network congestion, requiring renegotiation. If the actual QoS values in the system change over time, adjustments in the transport subsystem or in the operating system are initiated. Thus, the system must constantly monitor the actual QoS and employ correction mechanisms such as blocking lower priority tasks.

5.3 QoS in ATM Network

In the ATM world, QoS guarantees can be provided either pro-actively or reactively [24]. In reactive quality of service management schemes, the sources are allowed to send data in any fashion they choose. Resources are not allocated to individual connections. However, when the network manager senses that the quality of the service provided by the network is in danger, or likely to be in danger, it takes remedial action to correct this situation, possibly by requesting some sources to send data at a lower rate. However in wide area networks with high latencies, the network may not be able to react fast enough. Thus, proactive techniques involving resource management at call establishment time need to be explored.

Network resource management at call establishment comprises of the following framework. At call setup, a user declares the burstiness characteristics of the call to the network with call control parameters. Based on theses parameters, the network manger allocates the necessary resources to the given connection to assure the quality required for cell transfer delay and cell loss. If there are insufficient network resources available, the call will be rejected. The network monitors the cell stream coming from the user to verify that the stream conforms to the parameter values declared at call establishment. It also maintains a scheduling policy which allows more important or delay-sensitive cells to be given priority.

We can identify four crucial components of this resource management architecture [10, 11]:

• Specification : the method by which the burstiness and traffic characteristics of the communicating clients are specified;

• Rate Control : the mechanism used by the network to ensure that this contract is being followed by the clients;

• Scheduling Policy : determines which cell is to be transmitted next wherever queuing of cells may occur; and

• Admission Control : the set of rules which allocates resources and determines if it is possible to accept a new connection.

The provision of performance guarantees by means of resource allocation can only be done efficiently if the clients specify their traffic characteristics and obey them when transmitting data. The following qualities must be found in any specification:

• the specification must be exact, i.e. the network must be able to verify whether the client is obeying its promises or not;

• the specification must be flexible, i.e. the clients are allowed to specify traffic with widely differing characteristics of burstiness and bandwidth requirements;

• the specification must be simple and concise; and

• the specification must be maintainable.

The traffic characteristics of any connection will change as we move along the path of a connection. In general, the burstiness of any connection tends to increase as we proceed in this fashion. Thus resources allocated to a connection in this fashion will tend to increase along the path. The network must be able to account for this change in traffic specification and ideally be able to preserve the same traffic specification from node to node.

Three simple traffic specifications have been proposed in existing literature for admission control.

• The average rate model : [15] specifies the average bandwidth required by the connection over an averaging interval. This is like a window-based mechanism, where at most r packets can be sent in an averaging interval of length I. Alternately, we can look upon it as a rate- based specification, when the average spacing between two packets is at least x_{ave} in every interval of length I (with the obvious equivalence that $x_{ave} = I/r$). A slight variation of the scheme would specify that the spacing between any two packets be at least x_{ave}. The disadvantage of the average rate model is that it does not allow for burstiness in a client's traffic. Thus, this model must be augmented by models which allow for variable rates or bursts.

• The average-rate burst-size model: [8] specifies a maximum burst size in addition to an average rate. In every time interval of length I, the number of packets on the connection must not exceed $I/x_{ave} + b_{max}$, where x_{ave} is the average spacing between the packets and b_{max} is the maximum burst size. An intuitive explanation of the model is that it allows up to b_{max} packets to be transmitted back-to-back, but maintains a long-term average rate of I. This model allows some burstiness, reflected by the burst-size parameter. However, the burstiness (the burst size b_{max}) increases as one moves across the nodes in a network. In terms of resource allocation, this model tends to tax the nodes downstream much more heavily than the nodes upstream.

- The peak-rate average-rate model: [12] This model allows the sender to transmit at two different rates, a peak and an average rate. The minimum spacing between any two packets must be larger than x_{min} at all times (this corresponds to the peak rate); while the average packet spacing in any interval of length I must be x_{ave}. In this model, burstiness may be expressed as the ratio of peak to average bandwidth or x_{ave}/x_{min} [8]. This model allows burstiness to be modeled, and has the advantage that the minimum and average spacings can be maintained at all nodes along the connection's path by a simple rate-control mechanism. This has obvious benefits for resource allocation schemes at the nodes downstream.

In general, the traffic generated by the clients in a B-ISDN network will not be well-behaved and the client will need a leaky bucket mechanism at the entry of the network to convert the natural unregulated traffic into the regulated specified traffic adhering to the above mentioned models.

5.4 QoS in Legacy LAN

As mentioned earlier, the protocols for regular data transfer in legacy LANs do not have any resource reservation mechanism in order to guarantee a particular QoS for real-time multimedia traffic. Some possible solutions are presented in this section for networks with ethernet topologies that are connected to ATM backbones. These will be the most predominant installations found at various sites in a few years.

In this method a two priority feature is added to medium access protocol of the CSMA/CD protocol stack and is supported by message passing techniques. The two levels of priority: high priority (for real-time or multimedia traffic) and low priority (for traditional data traffic). A station wishing to send high-priority traffic must inform all other stations on its segment including the hub it is connected of its intention, by broadcasting a request message. The request message will indicate the session length for this transfer and hence for that period the other stations will not transmit any low priority traffic. The requesting station will have to permit for a time period that will be required to clear the bus of low priority traffic.

When stations receive a message indicating high priority traffic needs to be sent by a station on the same segment, they stop sending low priority traffic immediately. This is a preemptive mechanism and is in place for the time period of the high priority session. Multiple multimedia sessions may take place at the same time as the bus or segment is reserved for high priority traffic. Stations on the segment keep track of the number of stations transmitting real-time traffic. After a station completes transmission of its multimedia data, it broadcasts a completion message. This results in each station on the segment reducing its count of number of stations transmitting high priority traffic by one. It is important to understand that this completion message must be treated as a high priority message and must be received by all stations on the segment.

The transport medium i.e. ethernet segment may be time multiplexed between high priority and low priority traffic as shown in Figure 5.1. In the figure below, T_H is the time period used for high priority traffic (or multimedia) and T_L is the time period for low priority (or regular data) traffic. The period T_H starts when high priority traffic sessions begin and ends when the count of multimedia sessions is decremented to zero or if a predetermined maximum limit is reached. The maximum limit is used to ensure that the ratio T_H/T_L is fixed to ensure that starvation does not take place. The number of simultaneous multimedia sessions must be limited so that the number of collisions are minimized.

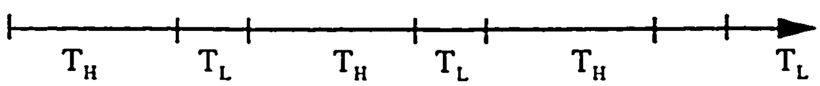


Figure 5.1 : Time Periods for Different Priorities of Traffic

This scheme may be extended to include multiplexing of the time available for each stations to transmit multimedia traffic during the time period T_H. All the stations on a segment use a fixed quantum T_Q for high priority traffic. A station uses the ethernet bus for the quantum of time and then places a request for another quantum after the expiration of the current quantum. Each station must maintain a table which contains the time of the latest transmission request from all the stations. An entry for a station is updated when a request message is received indicating that it wants to start its multimedia session. The station with the smallest timestamp is allocated the bus bandwidth after the current quantum expires. A station can find if the current quantum has expired by comparing the difference between the latest timestamps in its table and the current time, with the default session length quantum. If the current time minus the value of the latest timestamp is less than or equal to the value of T_Q, the station may proceed with its request immediately. If not, then the station tries whenever the current quantum expires.

5.5 QoS in a Heterogenous Network

If shared media hubs are replaced by switching hubs we are ensured of 10 Mb/s on each of the ports which represents a segment. Thus, the available bandwidth is increased for the number of stations hanging of a port. If these switching hubs are connected to the ATM backbone then the priority scheme in the LAN medium access protocol stack can be converted to the ATM specifications in ATM adaptation layers (i.e. AAL 1-5). During the period for multimedia traffic T_H the switching hub negotiates with the ATM backbone for CBR or VBR channels with a bandwidth of at least 10 Mb/s from itself to some other switching hub which provides connectivity for the recipient station. The limitation of 10 Mbps is due to the capacity of the stations' connection to the network. It must be noted that we are assuming that each workstation is connected directly to a port on the switching hub. If the switching ports are connected to shared medium hubs, which have the sending and receiving workstations hanging of its ports, then the bandwidth available to each workstation is less than 10 Mbps. If the low priority time period is being implemented then the channels on the ATM backbone are used with UBR specification. The scheme can receive further support if buffers are available for each type of traffic in the switching hubs. The buffer capacity for the high priority traffic is larger than the buffer for low priority traffic to accommodate for the end-to-end delay constraints.

5.6 QoS Negotiation in an ATM Network

If the communication support provided by the network is not a best-effort service but it is able to assure quality of service, such as in an ATM network, the general form of the interaction between the client and the network necessitates an establishment phase, in which the client requests the network for a connection characterized by some performance requirements and the network verifies whether this request can be accommodated. The steps in this interaction, if the there is no central network management entity, are as follows:

1. The client determines the values of the communication parameters characterizing a connection to be established with one or more destination nodes. It is assumed that the client does not have, in general, any previous information on the network state.

2. The network receives the request and evaluates it. This evaluation is not centralized in a particular node, but it is performed by submitting the request to each node on the path selected between the source and the destination. If a node has sufficient processing power and resources available to accept the connection without compromising the performance of the already existing channels, an adequate amount of resources are conditionally reserved for the requesting client.

3. The request proceeds toward the destination until one of the following situations arise:

- one of the nodes along the path cannot accommodate the new channel, or
- the destination is reached.

In the first case an answer is elaborated by that node stating the rejection of the request. In the second case, some final tests are executed at the destination.

4. Starting from the last node reached, an answer message proceeds to the requesting host, following the same path of the request message. If the answer contains a rejection message, the resources tentatively reserved are released, otherwise they are confirmed and the channel is established.

In general, it is assumed that the client should proceed following a converging algorithm. First the client sends a request for a channel which will best fit his traffic and performance requirements. Then the network replies with an accepted or rejected message. If the request has been refused, then the client has two possibilities: the first one is to repeat the same request until it will be accepted. In addition to generating a flood establishment related traffic on the network, this solution causes a waste of time and computing resources for the client, which periodically has to submit the request and for the network, which has to accommodate it. Since rejections of channel establishments are most likely to happen during periods of network congestion, this procedure seems to be particularly inappropriate.

The second possibility is that the client modifies the request, if this is feasible, by changing the value of some communication requirements and submits it again to the network. This solution appears to be more logical than the first one, provided the client receives some information from the network in the rejection message. The preparation of the second "guess" can take advantage of this information. In this method it is important to

```
node = origin
request_message[origin] = parameter_list

while (node != destination)
do
    request_message[node] =compute_local_test(request_message)
   node = succ(node)
done

request_message[destination] = compute_final_test(request_message)
return_message = prepare_message(request_message)

while node != origin
do
   node = prev(node)
   if return_message != OK    # at least one node test failed
          release_resources(node)
   else
          confirm_resources(node)
done
```

Figure 5.2 : Algorithm for Resource Reservation in an ATM Network

note that the channel establishment request must traverse from source to destination, even though rejections have been received from some nodes along the path. In the case a node is not able to accommodate the new channel, the reason for the rejection and the results of the local tests are included in the establishment message. When the message reaches the destination, the final tests are always performed. If these tests along with the ones previously conducted at the intermediate nodes yield positive answers, a "request accepted" message is prepared for the client. Otherwise, if there are negative answers, the rejecting nodes include their reasons for rejection in a rejection message to the client. A brief description of the algorithm is given in Figure 5.2.

To avoid the problem of competing again for resources that originally were available, the resources reserved during the rejected establishment message are not released immediately, but are kept available to the same client for a limited amount of time. This time has to be evaluated carefully, in order to avoid that these resources can be unavailable to other clients too long. This will also reduce the call blocking probability for the new channel.

5.7 QoS Negotiation in a Heterogenous Network

In a heterogenous network environment consisting of ethernet and ATM the negotiation can be extended to gather information about the type of network that will be traversed between source and destination. Consider that a station is connected to an ethernet network. This network is further connected to an ATM backbone and/or an ATM wide area network. This source station would like to communicate multimedia data to a destination.

The source station may follow the following steps:

1. Inform the ethernet hub to which it is connected of its intention to transmit multimedia data. This request will also contain the quality of service parameters that the station requests from the network.
2. If the network component encountered is not the destination and the topology is ethernet then the request is considered a message for high

priority traffic as discussed in the previous section. It must be noted that this connection will only offer a best-effort guarantee for the transfer of data.

```
node = origin
request_message[origin] = parameter_list
net_type = ethernet

while (node != destination)
do
     if network == ethernet
        node = succ (node)
    else
        request_message[node] = compute_local_test (request_message)
        net_type = hybrid
        node = succ (node)
done
if network == ethernet && net_type == ethernet
    return_message = OK
else
    if network == ethernet && net_type == hybrid
        return_message = prepare_message (request_message)
   else
        request_message[destination] =
compute_final_test(request_message)
        return_message = prepare_message (request_message)

while (node != origin)
do
        node = prev (node)
        if net_type == hybrid && return_message != OK
                release_resources (node)
        else
                confirm_resources (node)
done
```

Figure 5.3 : Algorithm for Resource Reservation in a Heterogenous Network

3. If the network component is an ATM interface then the request is considered for channel establishment and the quality of service parameters are tested at the interface to verify availability of resources.

4. The request will continue to its destination which may be ethernet or ATM connected. If the network traversed is purely ethernet then the request is never rejected. But if the network traversed is a combination of ethernet and ATM, a failure of the test at an ATM connected node will cause the request to be rejected. If the request is not rejected at any of the nodes, the resources at each of the nodes are reserved for the multimedia communication. The algorithm is described in Figure 5.3.

Thus, it is seen that with the above scheme we can establish a channel for multimedia communication which traverses multiple network topologies. This channel will be combination of best-effort and guaranteed quality of service for the communication requirements of the client application.

Chapter 6

Conclusion

6.1 Contributions

The research presented in this dissertation has focussed on the design of a high speed integrated network architecture for a distributed multimedia environment. This network architecture was simulated using a mathematical model and then implemented at the NASA Classroom of the Future Program. The multimedia server capacity was first calculated using a model and then implemented using the StarWorks architecture.

After implementation, a number of scenarios were monitored using the network for video conferences between clients, for distributing multimedia data objects from a video server. During these performance tests we found that for the network to provide a true multimedia transporting environment, a resource reservation scheme needs to be implemented as part of an operating system which is installed on the client workstations. The research also showed how the traffic on the network can be characterized for various load factors. This characterization was then used to determine the type of backbone network structure that needs to be put into place to carry multimedia communication.

Summarizing, the contributions of this dissertation are:

1. A new workload characterization to determine the capacity of an integrated network architecture for a distributed multimedia system.
2. A computational model for determining the structure of various components in the multimedia server architecture.
3. A paradigm for the integration of computer local are network and the video network based on legacy LAN and new switching technologies.
4. Established benchmarks for the operation of the network for client-server based multimedia applications and video conference broadcasts.
5. A foundation for the development of a framework for establishing channels between source and destination that can guarantee quality of service in networks consisting of legacy LAN and ATM technologies for multimedia communication.

6.2 Future Research

The research presented in this dissertation does not look into the implementation of the reservation scheme in the network designed at the NASA Classroom of the Future Program. Further, schemes for resource reservation within the application and the operating system of the workstation need to be addressed to guarantee that the frame rate for presentation is maintained at all times during the multimedia communication.

One of the key components of multimedia communication which was not studied in great detail in this research was the synchronization between the different components of multimedia streams and between different streams on the same client or the server. The whole notion event-based synchronization needs to be investigated in the environment constructed as part of this research.

The possibility of resource reservation in a wide area network is very complex, as the existence of a central management system cannot be taken for granted. This leads to the problem of how guaranteed quality of service can be obtained in a wide area network for a distributed multimedia system.

Bibliography

[1] J. F. Adam, H. Houh and D. Tennenhouse, *"A Network Architecture for Distributed Multimedia Systems,"* Proceedings of IEEE International Conference on Multimedia Computing and System, IEEE CS Press, Los Alamitos, California 1994, pp. 76-85.

[2] M. Alfano, *"A Quality of Service Management Architecture (QoSMA): A Preliminary Study,"* TR-95-070, International Computer Science Institute, Berkeley, December 1995.

[3] D. P. Anderson, *"Metascheduling for Continuous Media,"* ACM Transactions on Computer Systems, Vol. 11, No. 3, August 1993, pp. 226-252.

[4] R. T. Apteker, J. A. Fisher, V. S. Kisimov and H. Neishlos, *"Video Acceptability and Frame Rate,"* IEEE Multimedia, Fall 1995, pp. 32-40.

[5] ATM Forum, *"ATM: User-Network Interface Specification, Version 3.0,"* Prentice Hall, Englewood Cliffs, N.J. 1993.

[6] S. Boecking, et. al., *"The BERKOM Multimedia Transport Protocol,"* IS&T/SPIE Symposium on Electronic Imaging: Science and Technology, Proceedings High-Speed Networking and Multimedia Conference, Vol. 2188, SPIE, New York, February 1994, pp. 256-265.

[7] D.D. Clark, S. Shenker and L. Zhang, *"Supporting Real-Time Applications in an Integrated Services Packet Network: Architecture and Mechanism,"* Proc. SigComm 92, ACM, New York, 1992, pp. 14-26.

[8] R.L. Cruz, *"A Calculus for Network Delay and a Note on Topologies of Interconnection Networks"*, Report No. UILU-ENG_87-2246, University of Illinois, July 1987.

[9] D. Deloddere, W. Verbiest and H. Verhille, *"Interactive Video on Demand"*, IEEE Communications, Vol. 32, No. 5, May 1994, pp. 82-88.

[10] D. Ferrari, *"Guaranteed Performance in Wide-Area Networks"*, TR-89-022, International Computer Science Institute, Berkeley, June 1989.

[11] D. Ferrari and D. C. Verma, *"A Scheme for Real-Time Channel Establishment in Wide Area Networks"*, IEEE Journal on Selected Areas in Communication, Vol. 8, No. 3, April 1990, pp. 368-379.

[12] D. Ferrari, *"Client Requirements of Real-Time Communication Services,"* IEEE Communications, Vol. 28, No. 11, Nov. 1990, pp. 65-72.

[13] A. G. Fraser, *"Early Experiments with Asynchronous Time Division Networks,"* IEEE Network, Jan. 1993, pp. 10-12.

[14] R. Garg, *"Characterization of Video Traffic,"* TR-95007, International Computer Science Institute, Berkeley, January 1995.

[15] S. J. Golestanis, *"Congestion-Free Communication in Broadband Packet Networks"*, Proceedings ICC 1990, pp. 308.3.1-308.3.6.

[16] M. Hayter and D. McAuley, *"The Desk Area Network,"* ACM Operating System Review, Vol. 25, No. , Oct. 1991, pp. 14-21.

[17] M.E. Hodges et al, *"A Construction Set for Multimedia Applications"* IEEE Software, pp. 37-43, January 1989.

[18] A. Hopper, *"Pandora - An Experimental System for Multimedia Applications,"* ACM Operating System Review, Vol. 24, No. 2, Apr. 1990, pp. 19-34.

[19] R. Jefferies, *"ATM LAN Emulation: The Inside Story,"* Data Communications, September 1994, pp 95-100.

[20] J. Jung and D. Seret, *Translation of QoS Parameters into ATM Performance Parameters in B-ISDN,"* Proceedings of IEEE INFOCOMM '93, San Francisco, March 1993, pp. 748-755.

[21] J. Lane, *"Asynchronous Transfer Mode: Bandwidth for the Future,"* Telco Systems, 1992.

[22] A. A. Lazar and G. Pacifici, *"Control of Resources in Broadband Networks with QoS Guarantees,"* IEEE Communications Magazine, October 1991, pp. 66-73.

[23] D. LeGall, *"MPEG: A Video Compression Standard for Multimedia Applications"*, Communications of the ACM, April 1991.

[24] B. Leiner, *"Critical Issues in High Bandwidth Networking,"* Internet RFC 1077, November 1988.

[25] W.F. Leung et al, *"A Software Architecture for Workstations Supporting Multimedia Conferencing in Packet Switching Networks"*, IEEE Journal on Selected Areas in Communications, Vol. 8, No.3, pp. 380-390, April 1990.

[26] T.D.C. Little and A. Ghafoor, *"Network Considerations for Distributed Multimedia Object Composition and Communication"*, IEEE Network Magazine, pp. 32-49, November 1990.

[27] T.D.C. Little and A. Ghafoor, *"Synchronization and Storage Models for Multimedia Objects"*, IEEE Journal on Selected Areas in Communications, Vol. 8 No.3, pp. 413-427, April 1990.

[28] A.C. Luther, *"Digital Video in the PC Environment"*, McGraw-Hill, New York, 1989.

[29] M. Macedonia and D. Brutzman, *"MBONE, the Multicast Backbone"*, IEEE Computer, Vol.27, No. 4, April 1994, pp. 30-36.

[30] K. Maly, H. Abdel-Wahab, C.M. Overstreet, Ajay Gupta, Muthu Kumar and Rahul Srivatsava, *"Issues in Scaling Multimedia Collaboration Tools for Remote Instruction,"* IEEE Symposium on Computers and Communications, Alexandria, Egypt, June 27-29, 1995.

[31] S.E. Minzer, *"Broadband ISDN and Asynchronous Transfer Mode (ATM)"*, IEEE Communications Magazine, September, 1989.

[32] H.E. Meleis and D.N. Serpanos, *"Designing Communication Subsystems for High-Speed Networks"*, IEEE Network, July 1992, pp. 40-46.

[33] K.M. Nichols, *"Network Performance of Packet Video on a Local Area Network"*, Proceedings IEEE Phoenix Conference on Computers and Communications, April 1-3, 1992.

[34] C. Nicolaou, *"An Architecture for Real-Time Multimedia Communication Systems,"* IEEE Journal on Selected Areas in Communication, April 1990.

[35] Saurab Nog and David Kotz, *"A Performance Comparison of TCP/IP and MPI on FDDI, Fast Ethernet and Ethernet,"* Dartmouth College, PCS-TR95-273, November 1995.

[36] NSF, *"Research Priorities in Networking and Communications,"* NSF Division of Networking and Communications Research and Infrastructure, May 1994.

[37] G. M. Parulkar and J. S. Turner, *"Towards a Framework for High Speed Communications in a Heterogenous Networking Environment"*, Proceedings INFOCOMM '89, pp. 665-668.

[38] M. de Prycker, *"Asynchronous Transfer Mode: Solution for Broadband ISDN,"* Ellis Norwood, 1993.

[39] K. Ramkrishnan, P. Biswas and R. Karedia, *"Analysis of File I/O Traces in Commercial Computing Environments,"* Performance Evaluation Review, Vol. 20, No. 1, June 1992, pp. 78-90.

[40] V. P. Rangan and H. M. Vin, *"Designing File Systems for Digital Video and Audio,"* ACM Trans. Computing Systems, Vol. 18, No. 2, June 1992, pp. 197-222.

[41] V. P. Rangan, H. M. Vin and S. Ramanathan, *"Designing an On-Demand Multimedia Service,"* IEEE Communications, Vol. 3, No. 7, July 1992, pp. 56-64.

[42] M. Salmony and D. Shepherd, *"Extending OSI to Support Synchronization Required by Multimedia Applications"*, Computer Communication, Vol. 13, September, 1990, pp. 399-406.

[43] J. Schnepf, D. H. C. Du, E. R. Ritenour and A. J. Fahrmann *"Building Future Medical Education Environments over ATM Networks,"* Communications, ACM, Vol 38, No. 2, Feb. 1995.

[44] R. Simon, D. Krieger, T. Znati, R. Lofink and R. Sclabassi, "Multimedia MedNet: A Medical Collaboration and Consultation System," Computer, IEEE, Vol. 28, No. 5, pp 26-38, May 1995.

[45] J. Smith, E. Cooper and B. Davie, guest eds., special issue on *"High-Speed Computer/Network Interfaces,"* IEEE J. Selected Areas in Communications, Vol 11, No. 2, Feb. 1993.

[46] R. Steinmetz and K. Nahrstedt, *"Multimedia: Computing, Communications and Applications,"* Prentice Hall, New Jersey, 1995.

[47] H. J. Stuttgen, *"Network Evolution and Multimedia Communication,"* IEEE Multimedia, Fall 1995, pp. 42-59.

[48] A. S. Tanennbaum, *"Computer Networks,"* Second Edition, Englewood Cliffs, New Jersey, Prentice Hall, 1988.

[49] F. A. Tobagi and J. Pang, *"StarWorks-A Video Applications Server,"* Proceedings of IEEE Compcon Spring 93, CS Press, Los Alamitos, Calif., 1993, pp. 4-11.

[50] F. A. Tobagi, *"Streaming Raid - A disk Array Management System for Video Files,"* Proc. First ACM International Conf. Multimedia, ACM, New York, 1993, pp. 393-400.

[51] C. Topolcic, Editor, *"Experimental Internet Stream Protocol, Version 2 (ST-II),"* Internet RFC 1190, Oct. 1990.

[52] M. Woo, *"A Synchronization Framework for Communication of Pre-orchestrated Multimedia Information,"* IEEE Network, January/February 1994, pp. 52-61.

[53] S. Wray, T. Glauert and A. Hopper, *"Networked Multimedia:The Medusa Environment"* IEEE Multimedia, IEEE, Vol. 1, No. 4, pp. 54-63, 1994.

Vita

Nitin Naik obtained a bachelor of engineering degree from the University of Mysore in 1984. Thereafter, he was employed as an engineer at the Tata Electric Companies in Bombay. He came to the United States of America to study in the graduate program at Louisiana State University in the fall of 1986. He graduated with a master of science degree in electrical engineering in 1988 and a master of science degree in systems science in 1990.

He was employed with the College of Education at L.S.U. as a computer support analyst on the Louisiana Teacher Internship and Evaluation Programs from 1988 to 1991. In 1991, he was hired as the Technology Systems Manager at the College of Education, L.S.U. He collaborated with the schools in the surrounding area to establish connectivity between them and the college. Nitin is presently employed at the NASA Classroom of the Future Program in Wheeling, WV., where he has implemented a state-of-the-art computer-video local area network. This network is part of the research carried out for his doctoral dissertation and was nominated for the 1996 Computerworld Smithsonian Award for Innovation in Education and Academia. His doctor of philosophy degree will be conferred in August, 1996.

DOCTORAL EXAMINATION AND DISSERTATION REPORT

Candidate: NITIN S. NAIK

Major Field: Computer Science

Title of Dissertation: An Integrated Network Architecture for a High Speed Distributed Multimedia System

Approved:

Major Professor and Chairman

Dean of the Graduate School

EXAMINING COMMITTEE:

Date of Examination:

June 10, 1996